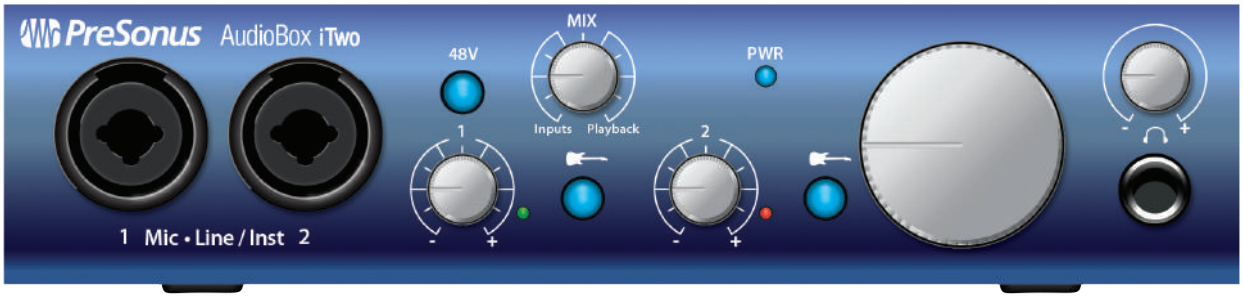
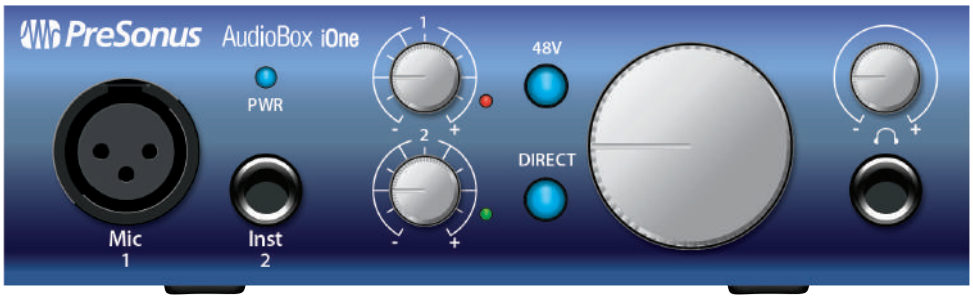
# AudioBox™ iOne and iTwo

## USB/iPad Recording Systems

Owner's Manual




PreSonus® www.presonus.com

English

# Table of Contents


## 1.0 Overview — 1

1.1 Introduction — 1

1.2 Summary of Hardware Features — 1

1.3 Summary of Included Software Features — 2

1.3.1 Studio One Artist — 2

1.3.2 Capture for iPad — 2

1.4 What's in the Box — 2

## 2.0 Hookup — 4

2.1 Front-Panel Controls and Connections — 4

2.2 Rear-Panel Connections — 6

2.3 Basic Hookup Diagrams — 8

2.3.1 iOne computer hookup diagram — 8

2.3.2 iOne iPad hookup diagram — 8

2.3.3 iTwo computer hookup diagram — 9

2.3.4 iTwo iPad hookup diagram — 9

## 3.0 Connecting to a Computer or iOS Device — 10

3.1 Minimum System Requirements — 10

3.2 Installation for Mac OS X and iOS — 10

3.3 Installation for Windows — 11

3.4 Windows PreSonus Driver Control Panel — 14

3.5 Using the AudioBox with Popular Audio Applications — 16

## 4.0 Studio One Artist Quick Start — 18

4.1 Installation and Authorization — 18

4.2 Enabling the Audio Driver — 19

4.3 Configuring MIDI Devices (AudioBox iTwo) — 20

4.4 Creating a New Song — 23

4.5 Creating Audio Tracks — 26

4.6 Anatomy of an Audio Track — 27

4.7 Adding Virtual Instruments and Plug-in Effects to Your Song — 27

4.7.1 Drag-and-Drop Virtual Instruments — 28

4.7.2 Drag-and-Drop Effects — 28

## 5.0 Tutorials — 29

5.1 Microphone Types — 29

5.1.1 Condenser — 29

5.1.2 Dynamic — 29

5.1.3 USB Microphones and Other Types — 29

5.1.4 Stereo Microphone Placement — 29

5.2 A Brief Tutorial on Dynamics Processing — 32

5.2.1 Common Questions Regarding Dynamics Processing — 32

5.2.2 Types of Dynamic Processing — 33

5.2.3 Compression Setting Suggestions — 37

5.3 Equalizers — 39

5.3.1 What is an EQ? — 39

5.3.2 Equalization Settings: How to Find the Best and Leave the Rest — 41

5.4 EQ Frequency Guides — 42

5.5 EQ Setting Suggestions — 43

## 6.0 Technical Information — 46

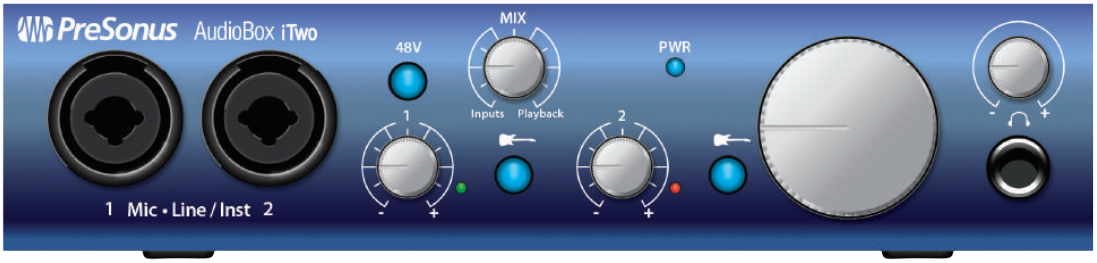
6.1 iOne and iTwo Technical Specifications — 46

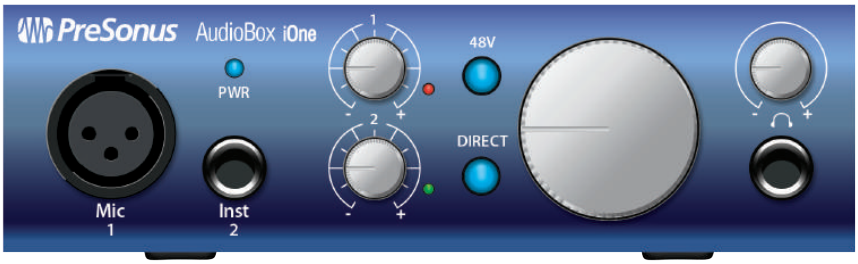
## 7.0 Troubleshooting and Warranty — 47

7.1 Troubleshooting — 47

7.2 Warranty — 48

# 1.0 Overview




## 1.1 Introduction

**Thank you** for purchasing a PreSonus AudioBox™ interface. PreSonus Audio Electronics, Inc., has designed the AudioBox iOne and AudioBox iTwo utilizing high-grade components to ensure long-lasting, optimum performance. Loaded with high-headroom, Class A microphone preamplifiers; a built-in 2x2 USB 2.0 and iPad recording and playback engine; MIDI I/O (AudioBox iTwo); and more, the iOne and iTwo extend the boundaries for musical performance and production.

All you need is a computer with a USB connection, a few microphones and cables, a pair of headphones or powered speakers, and your instruments, and you're ready to record in the studio or perform in front of a live audience! Both interfaces are MFi (Made For iOS) -compliant, so you can connect them to your iPad™ using the Device connection on the back panel. See iOS Devices for minimum system requirements.

PreSonus Audio Electronics, Inc., is committed to constant product improvement, and we value your suggestions highly. We believe the best way to achieve our goal of constant product improvement is by listening to the real experts: our valued customers. We appreciate the support you have shown us through the purchase of this product and are confident that you will enjoy your AudioBox!

## 1.2 Summary of Hardware Features

- 2x2 USB 2.0/iPad audio interface
- 24-bit/96 kHz sampling rate
- Class A microphone preamplifiers
- 48V phantom power
- Instrument inputs
- 2 line inputs (AudioBox iTwo)
- MIDI I/O (AudioBox iTwo)
- Headphone output with independent level control
- Zero-latency direct analog monitoring
- Rugged construction
- Complete mobile and studio solution with Studio One® Artist and Capture™ Duo for iPad
- Compatible with most Mac, Windows, and iOS audio software
- Mac® OS X, iOS, and Windows®-compatible ***(See Section 3.1 - Minimum System Requirements.)***

## 1.3 Summary of Included Software Features

### 1.3.1 Studio One Artist

The AudioBox iOne and iTwo interfaces include PreSonus Studio One Artist recording software (available for download), which comes with loads of plug-ins, loops, and samples, giving you everything you need for music recording and production.

The Studio One Artist Quick Start Guide and User Manual can be found in the help section of the software or in your My PreSonus account.

- Wirelessly transfer and import Sessions created in Capture for iPad®
- Unlimited track count, inserts, and sends
- 20 high-quality Native Effects™ plug-ins: amp modeling (Ampire), delay (Analog Delay, Beat Delay), distortion (RedlightDist™), dynamics processing (Channel Strip, Compressor, Gate, Expander, Limiter, Tricomp™), equalizer (Channel Strip, Pro EQ), modulation (Autofilter, Chorus, Flange, Phaser, X-Trem), reverb (MixVerb™, Room Reverb), and utility (Binaural Pan, Mixtool, Phase Meter, Spectrum Meter, Tuner)
- More than 6 GB of loops, samples, and instruments, featuring: Presence™ virtual sample player, Impact™ virtual drum machine, SampleOne™ virtual sampler, Mojito virtual analog-modeled subtractive synthesizer
- Project templates specifically for AudioBox interfaces
- Innovative MIDI mapping
- Powerful drag-and-drop functionality for faster workflow
- Mac OS X- and Windows-compatible ***(See Section 3.1 - Minimum System Requirements.)***

### 1.3.2 Capture for iPad

PreSonus Capture for iPad is a new mobile-recording application that is available from the Apple App Store.

Capture Duo is a free 2-channel app; Capture for iPad is a paid app that can record and play up to 32 channels. Other than the track count, all features are the same.

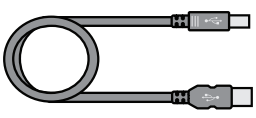
***See iOS Devices for minimum system requirements***.

- Simple, intuitive, mobile solution for easily capturing multitrack audio on your iPad
- Direct wireless transfer of sessions to Studio One Artist
- Built-in metronome
- Editing functions including cut, copy, paste, split, and duplicate
- Level and Pan controls for balancing your tracks

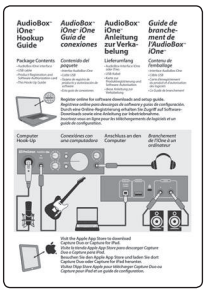
## 1.4 What's in the Box

Your AudioBox Package Contains the Following:

- An AudioBox iOne or AudioBox iTwo, depending on which you've purchased



- 5' (1.5m) USB cable




- AudioBox Hookup Guide

- Product registration card

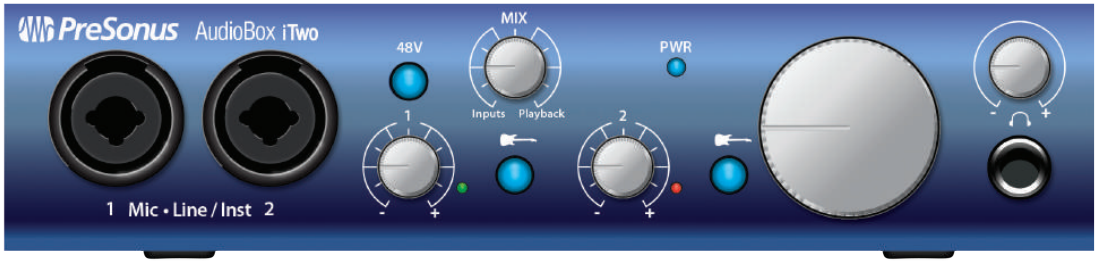
A complete software bundle is included to help you get the most from your AudioBox interface.

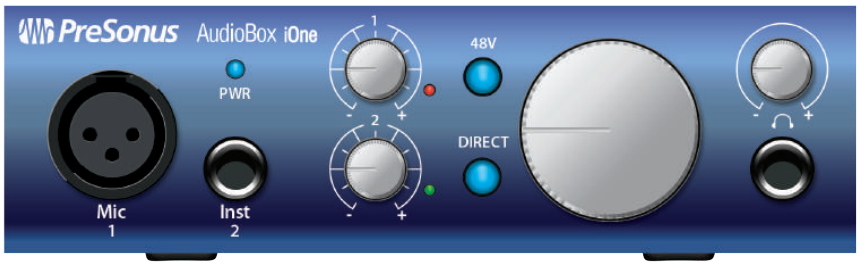
Software and services that complete your AudioBox iOne or iTwo solution include the following:

- Studio One Artist – Available for download from your My PreSonus account as soon as you register your AudioBox interface
- Capture Duo for iPad – A free download from the Apple App Store
- Nimbit® – A free account that's set up with your My PreSonus account so you can directly upload your recordings from Studio One Artist to sell or share with fans

# 2.0 Hookup

## 2.1 Front-Panel Controls and Connections



**Microphone/Instrument/Line Inputs.** Both channels of the AudioBox iTwo have Mic-Line/Instrument combo jacks that accept a ¼-inch TRS or XLR plug.

On the AudioBox iOne, Channel 1 offers an XLR jack, and Channel 2 has a ¼-inch Instrument jack.

**Please Note:** As with any audio input device, plugging in a microphone or instrument, or turning phantom power on or off, will create a momentary spike in the audio output of your AudioBox. Because of this, we highly recommend that you turn the input level all the way down before changing connections or turning phantom power on or off. This simple step will add years to the life of your audio equipment.

**Microphone Inputs.** These XLR inputs are equipped with high-headroom PreSonus microphone preamplifiers. These preamps work great with all types of microphones including dynamic, ribbon, and condenser microphones.

***Power User Tip:*** *Dynamic microphones and ribbon microphones (which are a special type of dynamic mic) are generally lower-output devices that, with few exceptions, require no external power source. Sending phantom power to a ribbon mic that doesn't require it can cause severe damage to the mic—usually beyond repair. Condenser microphones are generally more sensitive than dynamic and ribbon microphones and typically require external +48V phantom power. Always review your microphone's documentation and follow its recommended operating practices.*

**Instrument Inputs.** These ¼-inch connectors are for use with instruments that don't have internal preamps (guitar, bass, etc.). See Input Source Button for more information.

***Power User Tip:*** *Active instruments are those that have an internal preamp and a line-level output. This type of instrument should be plugged into a line input, rather than an instrument input. Plugging a line-level source into the instrument inputs not only risks damage to these inputs but also results in a very loud and often distorted audio signal.*

**Line Inputs.** The ¼-inch connectors on the AudioBox iTwo can also accept line-level inputs such as keyboards and samplers. See Input Source Button for more information.

**Input Source Button (AudioBox iTwo).** Each AudioBox iTwo input channel offers an Input Source button that lets you select instrument or line level for the ¼-inch inputs. When the button is not illuminated, the input will accept a line-level source, like a synthesizer or guitar amp modeler. Press this button to engage the instrument preamp when connecting guitars or a passive bass.

**48-volt Phantom Power.** The AudioBox iOne, and iTwo provide 48V phantom power for the microphone inputs. Pressing the 48V button switches phantom power on and off for the microphone inputs, and it will illuminate in blue whenever this function is switched on.

**WARNING:** Phantom power is only required for condenser microphones and can severely damage some dynamic mics, especially ribbon mics. Therefore, switch phantom power off when it is not required.

**XLR connector wiring for phantom power:**
**Pin 1** = GND **Pin 2** = +48V **Pin 3** = +48V

**Input Gain Control.** These knobs provide 52 dB of variable gain (-10 to +42 dB) on the microphone inputs and 51 dB of variable gain (-24 to +27 dB) on the instrument inputs.

**Input Signal Indicators.** Each channel features an LED next to the trim control that illuminates in green whenever an input signal is present. The color of the LEDs will change to red when the channel's input signal reaches -0.5 dBFS. At 0 dBFS your mic preamp/line signal will exhibit signs of clipping (distortion).

***Power User Tip:*** *Never run your input levels higher than the channel inputs can handle. If you overdrive the analog-to-digital converters, it will cause digital distortion (digital clipping), which sounds terrible.*

**Direct Monitor Mix Control (AudioBox iTwo).** The Mix knob allows you to blend your input signals with the playback streams from your computer. This allows you to monitor your input with zero latency. If the knob is positioned at 12 o'clock, the input and playback will be equally balanced.

Turning the knob to the left increases the level of the input relative to the playback; turning it to the right increases the level of the playback relative to the input.

**Please Note:** When creating monitor mixes using Studio One or another DAW, it's important that you turn the Mix knob all the way to the Playback position. Monitoring both the input and the playback will create a doubling effect that makes monitoring difficult.

**Direct Monitor Mix Control (AudioBox iOne).** The Direct button blends the source of what is heard through the Headphone and Main outputs in the following manner:

- In the Off, or "up," position (default), you'll only hear the playback from your computer.
- When the button is in the On, or "down," position, the button will illuminate in blue, and you'll hear a 50/50 blend of the playback from your computer and the input signal going into the interface.

**Headphone Output.** This ¼-inch TRS connector is where you connect headphones to your AudioBox interface.

**Headphone Knob.** This knob controls the volume of the headphone. The headphone amplifier is quite powerful, and the volume goes to 11, so use the maximum setting with extreme caution.

**Main Output Knob.** This knob controls the signal level for the Main outputs on the back of the interface.

**Power LED.** This LED illuminates in blue when the interface is properly powered and synced to a connection on you computer or iPad.

**Please Note:** If your AudioBox interface isn't properly synced, the LED will continue flashing blue until proper synchronization is achieved.

## 2.2 Rear-Panel Connections

**Kensington Security Slot.** This connector can be used with a Kensington®-style security lock to protect your AudioBox iOne or iTwo from possible theft.

**USB 2.0/Power Port.** This is for connecting your interface to an available USB port on your computer with the included USB cable. Because the AudioBox iOne and iTwo are bus-powered via this connection, they don't require an external power supply.

This port also provides power to the interface when used with an iPad. Simply use the included USB cable connected to a compatible 5V, 2.1A (2,100 mA) USB power supply, such as the one from an iPad.

***Power User Tip:*** *AudioBox iOne and iTwo are compatible with all USB 2.0 and 3.0 ports.*

**Device Port.** This lets you use an Apple-compatible 30-pin or Lightning-to-USB cable to connect your AudioBox to an iPad when recording and playing back from audio applications. (We recommend using the cable included with the iPad, as results may vary with non-Apple cables.) See iOS Devices for minimum system requirements.

**MIDI I/O (AudioBox iTwo).** MIDI stands for "Musical Instrument Digital Interface." MIDI inputs and outputs allow connection to, and communication with, external MIDI equipment. One function of these ports is MIDI sequencing but the MIDI protocol can be used for much more than instruments and sequencing.

***Power User Tip:*** *MIDI is not audio but is frequently used to trigger or control an audio source (such as synthesizer, keyboard, or sound module). It's important to ensure that your MIDI data is correctly sent and received by the appropriate hardware or software devices.*

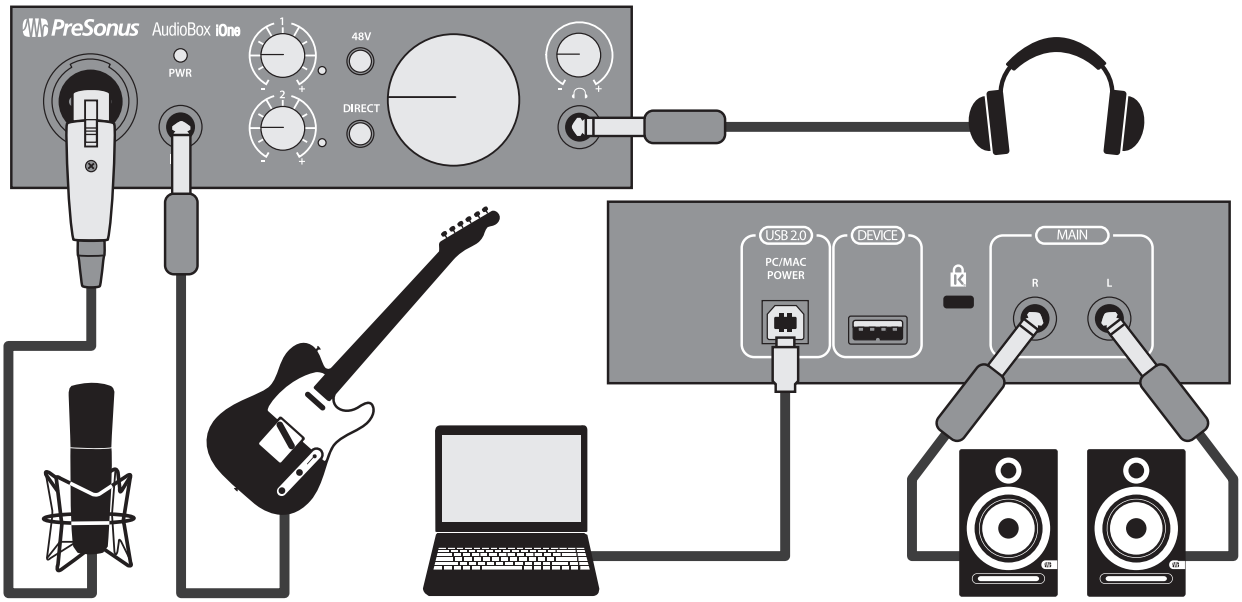
*If you want to record the audio from the device your MIDI Out is controlling, you'll need to return the audio to an AudioBox input channel. Please refer to the documentation provided for your MIDI devices for help with MIDI setup and usage.*

**Main Outputs.** These are the main outputs for your AudioBox interface. The Main Output knob on the front of the unit controls the volume level of these outputs.

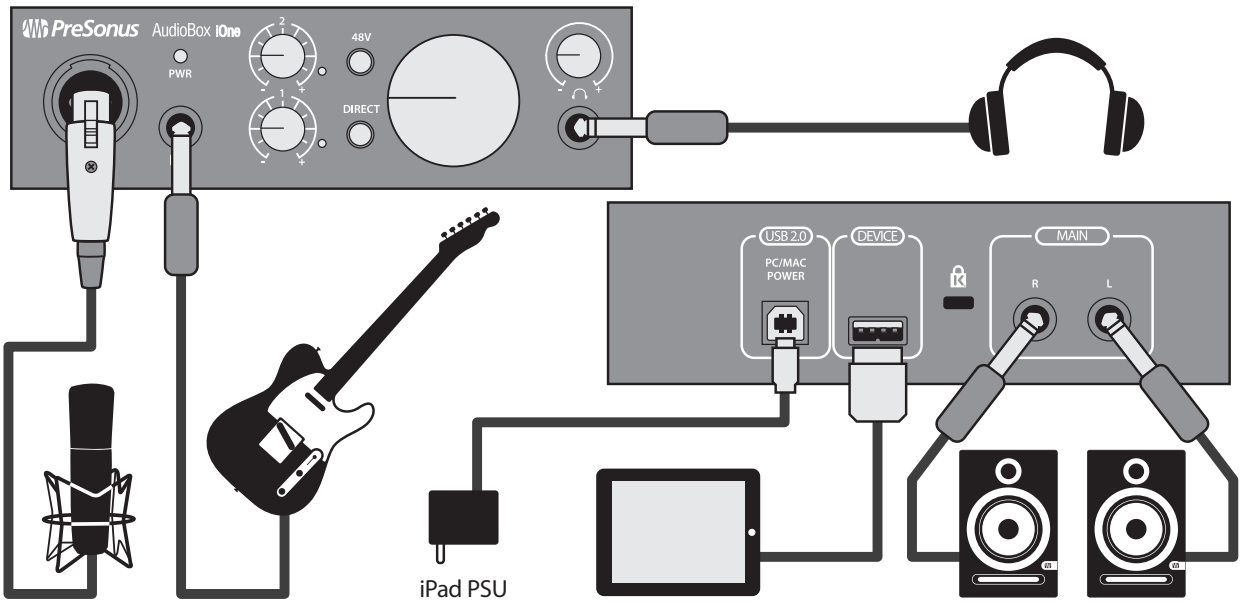
***Power User Tip:*** *You must connect speakers or headphones directly to your AudioBox interface in order to monitor playback from your computer.*

## 2.3 Basic Hookup Diagrams

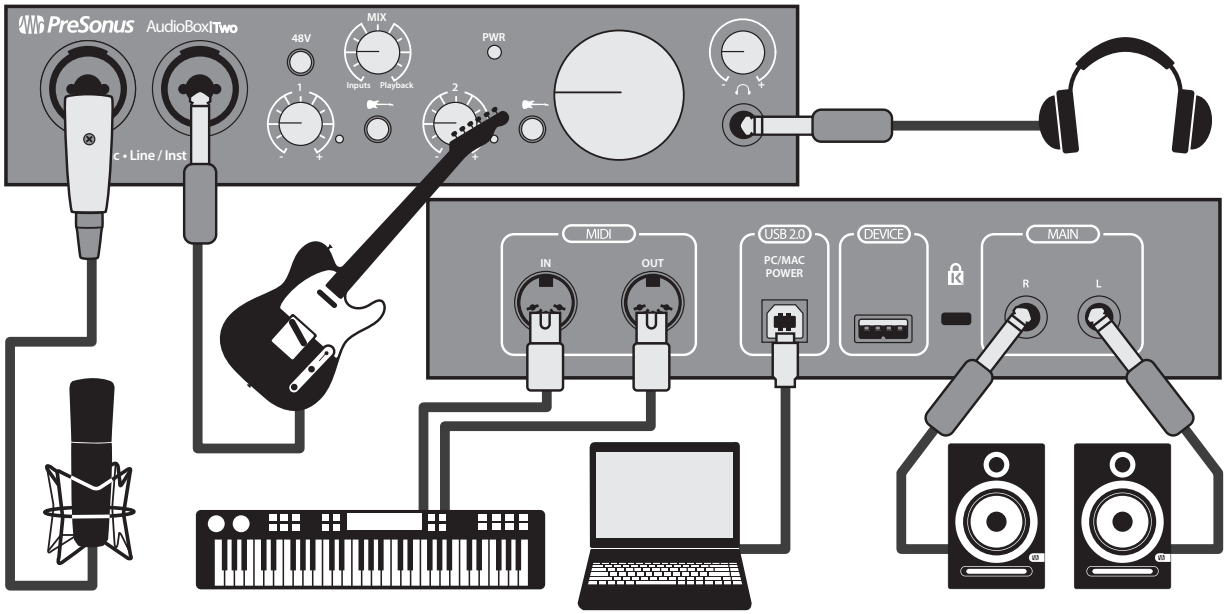
### 2.3.1 iOne computer hookup diagram



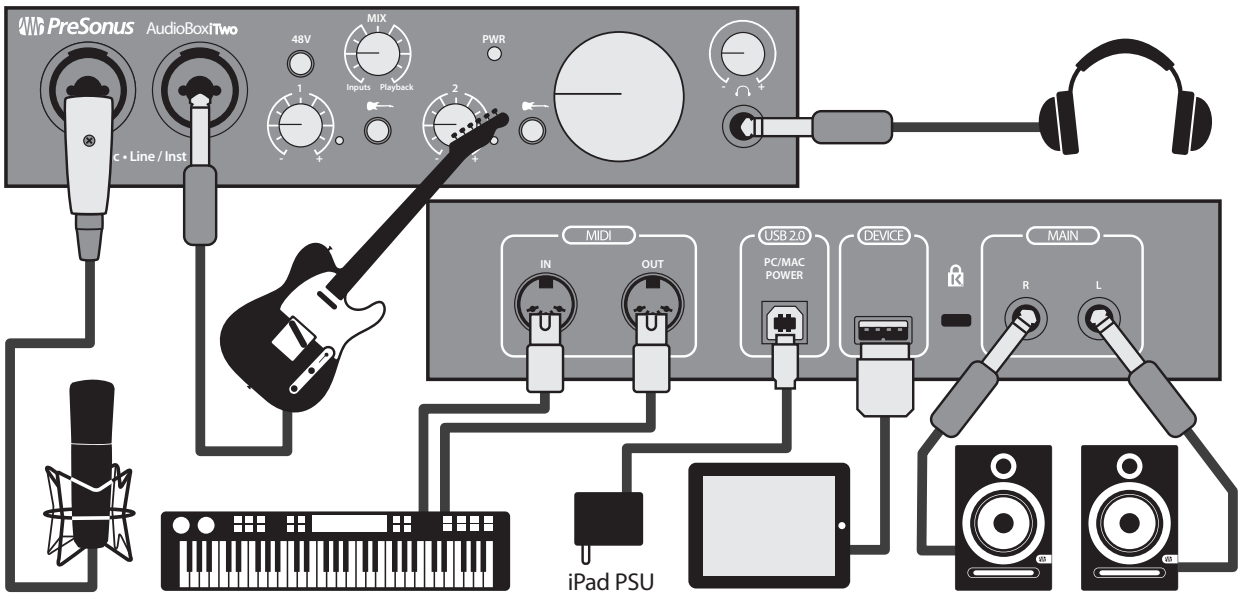

### 2.3.2 iOne iPad hookup diagram



### 2.3.3 iTwo computer hookup diagram




### 2.3.4 iTwo iPad hookup diagram



# 3.0 Connecting to a Computer or iOS Device

This chapter covers the minimum system requirements and instructions for connecting your AudioBox iOne or iTwo interface to your computer or compatible iOS device.

## 3.1 Minimum System Requirements

**Mac**

- Mac OS X 10.8 or later
- Intel Core 2 Duo (Intel Core i3 or better recommended)
- 4 GB of RAM (8 GB or more recommended)

**iOS Devices**

- iOS 7.03 or later
- iPad Air
- iPad mini with Retina display
- iPad (4th generation)
- iPad mini
- iPad (3rd generation)
- iPad 2

**Windows**

- Windows 7 x64/x86 SP1 or Windows 8 x64/x86 or later
- Intel Core 2 Duo (Intel Core i3 or better recommended)
- 4 GB of RAM (8 GB or more recommended)

**Please Note:** Minimum System Requirements are subject to change. Please visit www.presonus.com for the latest system requirements.

## 3.2 Installation for Mac OS X and iOS

**Basic Installation**

Because AudioBox iOne and iTwo are class-compliant Core Audio devices, there's nothing to install. Simply connect your interface to your Mac or iOS device to begin using it. A firmware update application is available but it is only required in the event a firmware update is released.

**Optional Configuration for Mac OS X System Sound Playback and/or Recording**

You can also set up your AudioBox interface as the default Input and Output device in OS X by following the steps listed below:

1. Open System Preferences by clicking the Apple menu. (System Preferences may also be located in Launch Pad, the Applications folder, and the Dock.)
2. Click "Sound" to open the Sound configuration.
3. On the Output tab, select "PreSonus AudioBox iOne" or "iTwo."
4. On the Input tab, select the "PreSonus AudioBox iOne" or "iTwo."
5. On the Sound Effects tab, select the "PreSonus AudioBox iOne" or "iTwo."
6. Close System Preferences.

***Power User Tip:*** *If you change the default sound settings on your computer, the sound will come through the speakers or headphones connected to the AudioBox iOne or iTwo interface instead of the internal computer speakers.*

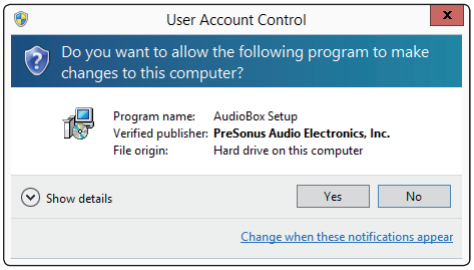
## 3.3 Installation for Windows

### Basic Installation

The AudioBox installer requires that your interface be connected to an available USB port on your PC. Once you've downloaded the installer, double-click the icon to launch the installer, which will take you through each step of the process. Please read each message carefully.

**Please Note:** It's recommended that you quit all applications and disable your anti-virus software, before starting the installation.

If you see a Windows Account Security alert asking if you want to allow the installer to make changes to your computer, click "Yes."


User Account Control
Do you want to allow the following program to make changes to this computer?
Program name: AudioBox Setup
Verified publisher: PreSonus Audio Electronics, Inc.
File origin: Hard drive on this computer
Show details
Yes
No
Change when these notifications appear

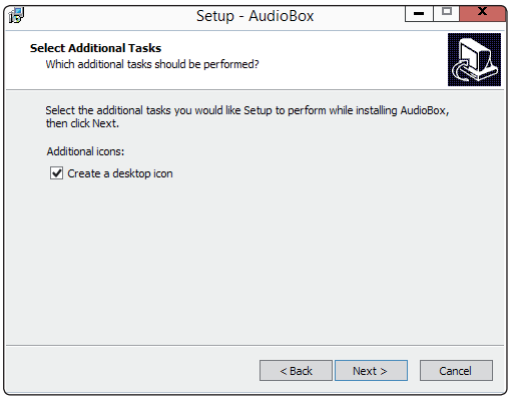
1. The installer will open to the Welcome screen. Click "Next."

Setup - AudioBox
Welcome to the AudioBox Setup Wizard
This will install AudioBox version 1.3 on your computer.
It is recommended that you close all other applications before continuing.
Click Next to continue, or Cancel to exit Setup.
Next >
Cancel

2. You will be asked if you would like the installer to automatically create an icon on your desktop for the AudioBox control panel.

   This control panel allows you to adjust the buffer size and sample rate of your AudioBox. Creating a desktop icon will provide you with quick access to these controls.

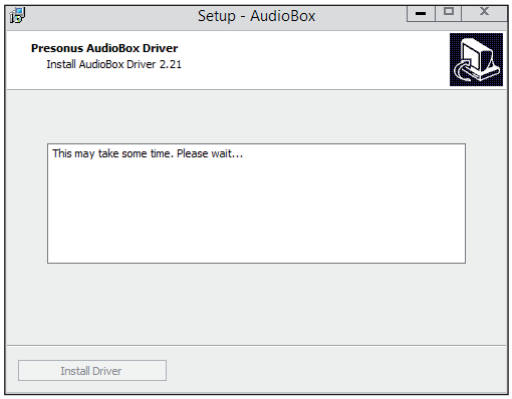
   Click "Next."


Setup - AudioBox
Select Additional Tasks
Which additional tasks should be performed?
Select the additional tasks you would like Setup to perform while installing AudioBox, then click Next.
Additional icons:
Create a desktop icon
< Back
Next >
Cancel

3. The installer will now scan your computer, and you'll be prompted to begin the installation.

   You'll be prompted to connect your interface if the installer doesn't detect it. However, you may need to wait a few seconds for the interface to appear.

   Click "Install Driver."



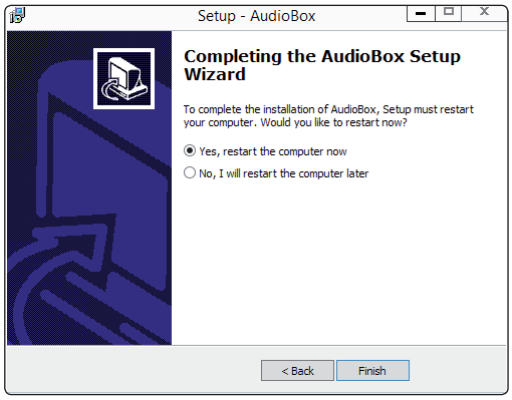

4. Once the installation has successfully completed, you'll be prompted to restart the computer.

   Click "Finish" to automatically restart your computer

**or**

   Click "Later" to close the installer and restart later.

**Please Note:** The driver will not function correctly unless your computer has been restarted.

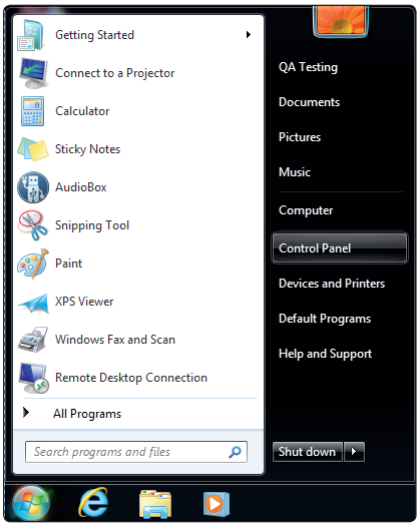

Setup - AudioBox
Completing the AudioBox Setup Wizard
To complete the installation of AudioBox, Setup must restart your computer. Would you like to restart now?
Yes, restart the computer now
No, I will restart the computer later
< Back
Finish

### Optional Configuration for Windows System Sound Playback and/or Recording

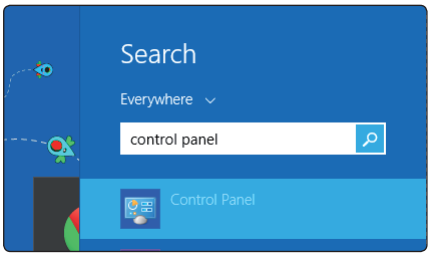
If your recording application doesn't support ASIO or WDM interfaces, you can set up your AudioBox interface as the Windows Default Playback and Recording device by following the steps listed below:

1. Open Control Panel.

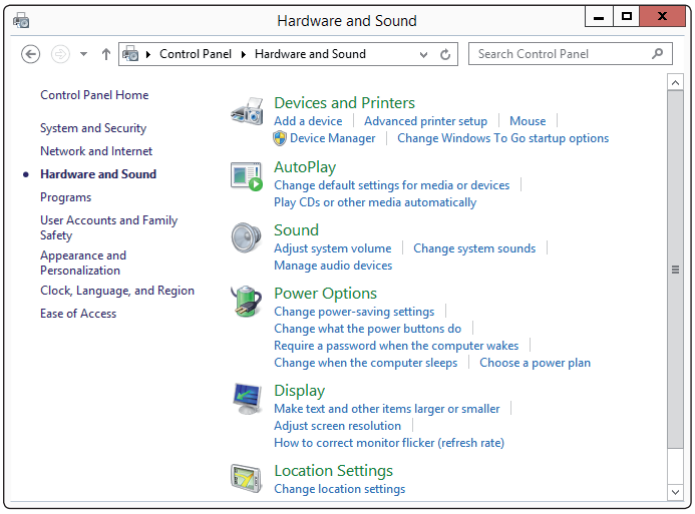
   In Windows 7, click Start>Control Panel.



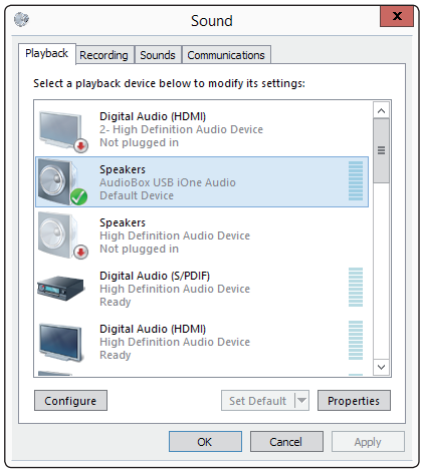

   For Windows 8+: From the Start Page, begin typing Control Panel and click the Control Panel icon when it appears.



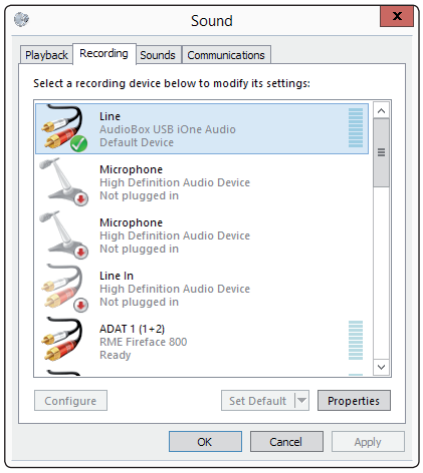

2. Click "Sound" to open the Sound-configuration dialog.


Hardware and Sound
Control Panel ▸ Hardware and Sound
Search Control Panel
Control Panel Home
System and Security
Network and Internet
Hardware and Sound
Programs
User Accounts and Family Safety
Appearance and Personalization
Clock, Language, and Region
Ease of Access
Devices and Printers
Add a device | Advanced printer setup | Mouse
Device Manager | Change Windows To Go startup options
AutoPlay
Change default settings for media or devices
Play CDs or other media automatically
Sound
Adjust system volume | Change system sounds
Manage audio devices
Power Options
Change power-saving settings
Change what the power buttons do
Require a password when the computer wakes
Change when the computer sleeps | Choose a power plan
Display
Make text and other items larger or smaller
Adjust screen resolution
How to correct monitor flicker (refresh rate)
Location Settings
Change location settings

3. On the Playback tab, select "Speakers - PreSonus AudioBox iOne" or "iTwo" and click "Set Default."




4. On the Recording tab, select the "PreSonus AudioBox iOne" or "iTwo" and click "Set Default."




5. Click "OK."

***Power User Tip:*** *If you change the default sound settings on your computer, the sound will come through the speakers or headphones connected to the AudioBox iOne or iTwo interface instead of the internal computer speakers.*

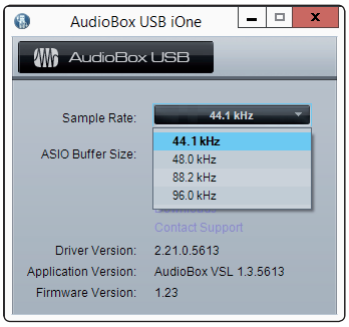
## 3.4 Windows PreSonus Driver Control Panel

When used with Windows systems, AudioBox iOne and AudioBox iTwo offer a separate control panel with the following settings:

- **Run at Startup.** When this is enabled, the AudioBox Control Panel will automatically launch each time your computer starts up.

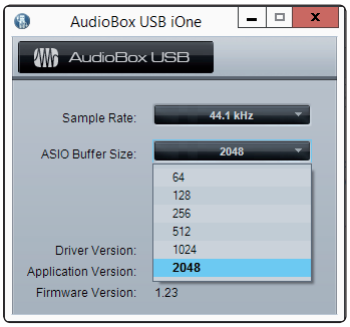
- **Sample Rate Selector.** Lets you set the sample rate to 44.1, 48, 88.2, or 96 kHz.

Although a higher sample rate results in better recording quality, it does increase the file size and the amount of system resources needed to process the audio.




***Power User Tip:*** *Changing the sample rate will produce a momentary popping noise. Because of this, it is recommended that you turn the Main and Phones volumes down on your AudioBox prior to changing the sample rate.*

- **ASIO Buffer Size.** Allows you set the buffer size from 64 to 2,048 samples.




***Power User Tip:*** *You should set your buffer size prior to launching your host application. Setting your buffer size to 64 or 128 may have a performance effect on your computer.*

*The buffer size determines the roundtrip time it takes audio data to be converted from analog to digital and back to analog. As a general rule, the higher the buffer size, the better the system performance, but the less playable virtual instruments and the like become.*

*In general, 512 samples (11 to 12 milliseconds) will provide you with a large enough buffer for optimum system performance but small enough to minimize latency and be unobtrusive.*

## 3.5 Using the AudioBox with Popular Audio Applications

Complete setup instructions for Studio One Artist and a brief tutorial on its features can be found in chapter 4 of this manual. However, you can use your AudioBox iOne or iTwo with almost any audio-recording application that supports Core Audio (Mac OS X or iOS) or ASIO (Windows).

Although this chapter covers basic driver-setup instructions for four popular audio applications, please consult the documentation provided with your software for specific instructions on how to select the correct AudioBox driver as the audio-device driver for your software.

**Apple Logic Pro/Express 7+**

1. Launch Logic Pro/Express.
2. Go to Logic | Preferences | Audio.
3. Click the Devices tab.
4. On the Core Audio tab, check Enabled.

   Logic Pro X doesn't have a Core Audio tab, but make sure Core Audio is enabled. The checkbox will be blue with a white checkmark in it.
5. Depending on which interface you have, select PreSonus AudioBox iOne or PreSonus AudioBox iTwo from the Output Device and Input Device menus.
6. Click the Apply Changes button.

   In Logic Pro X, Logic's Core Audio functionality will reinitialize on its own as soon as you click the Apply Changes button.
7. You will be asked if you'd like to relaunch Logic. Click "try (re) launch."
8. Close the Preferences window.
9. Your AudioBox features custom I/O labels for faster workflow.

   To enable these labels for use in Logic:
   - Go to Options | Audio | I/O Labels.
   - In Logic Pro X, open a session file and go to Mix | I/O Labels, or from within the Mixer window go to Options | I/O Labels.
   - The second column in the window is named "Provided by Driver." Activate each of these labels for your AudioBox. When you are done, close this window.
10. You're now ready to use your AudioBox interface.

**Steinberg Cubase 6+**

1. Launch Cubase.
2. Go to Devices | Device Setup.
3. Select "VST Audio System" from the Devices column in the Device Setup.
4. Depending on which interface you have, select PreSonus AudioBox iOne or PreSonus AudioBox iTwo from the ASIO Driver drop-down list.
5. Click "Switch" to begin using the AudioBox driver.
6. Once you've successfully changed the driver, go to Devices | VST Connections to enable your input and output buses.
7. You're now ready to use your AudioBox interface.

**Cakewalk Sonar 8.5+**

1. Launch Sonar.
2. Go to Options | Audio... and click on the Advanced tab.
3. Change the Driver Mode to "ASIO." (Using WDM, rather than ASIO, for pro-audio applications is not recommended.)
4. Click the "OK" button.
5. Restart Sonar.
6. Go to Options | Audio... and click on the Drivers tab.
7. Depending on which interface you have, highlight all input and output drivers for "PreSonus AudioBox iOne" or "PreSonus AudioBox iTwo."
8. Go to Options | Audio... and click on the General tab.
9. Based on the interface you have, set the Playback Timing Master to "PreSonus AudioBox ... DAW Out 1."
10. Based on the interface you have, set the Recording Timing Master to "PreSonus AudioBox ... Mic/Inst 1."
11. You're now ready to use your AudioBox interface.

**Ableton Live 8+**

1. Launch Ableton Live
2. Go to Options | Preferences | Audio
3. Choose Driver Type: ASIO | Audio Device: ASIO PreSonus AudioBox iOne or iTwo based on the interface you have.
4. Go to Input Config. Enable and select the desired Input channels.
5. Go to Output Config. Enable and select the desired Output channels.
6. You may now select the AudioBox inputs and outputs for each track created in Live.
7. You're now ready to use your AudioBox interface.

# 4.0 Studio One Artist Quick Start

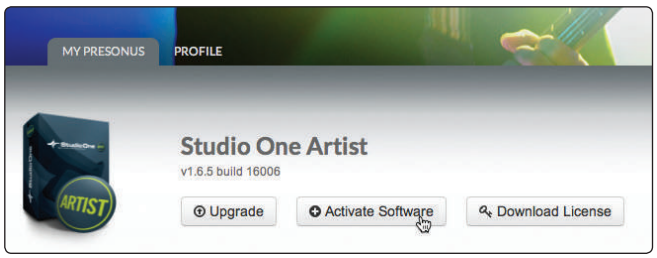
When Studio One Artist is launched, by default, you will be taken to the Start page. Here you'll find document-management and device-configuration controls, as well as a customizable artist profile, a newsfeed, and links to demos and tutorials from PreSonus. If you have an Internet connection on your computer, these links will be updated as new tutorials become available on the PreSonus Web site.

Complete information on all aspects of Studio One Artist can be found in the Help menu, or as a download from your My PreSonus account. The information in this tutorial covers only the basic aspects of Studio One Artist and is intended to get you set up and recording as quickly as possible.

## 4.1 Installation and Authorization

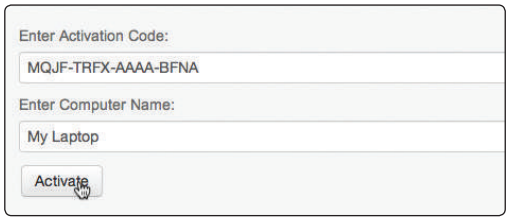
### Installing Studio One Artist

To install Studio One Artist, download the Studio One Artist installer from your My PreSonus account.




- **Windows users:** Launch the Studio One Artist installer and follow the onscreen instructions.
- **Mac users:** Drag the Studio One Artist application into the Applications folder on your Mac hard drive.

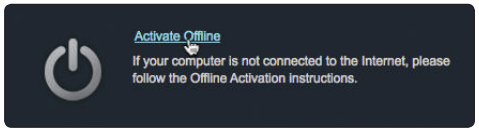
### Activating Studio One Artist Online




After installing Studio One Artist, launch the program, and the Activate Studio One menu will appear. Click on the Activate or Upgrade link to begin.

1. Click on the Activate Online link and enter the username and password you created during registration, along with the product key found in your My PreSonus account.
2. Click on the Activate button to finish the activation process.

### Activating Studio One Artist Offline


Activate Offline
If your computer is not connected to the Internet, please follow the Offline Activation instructions.

1. If your computer is not connected to the Internet, visit http://my.presonus.com on an Internet-connected computer to create your account.
2. Log in and click the "My Software" link.

3. Click on the "Download License" link.

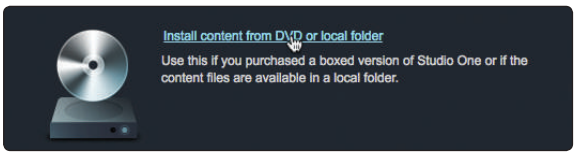
   A file called "Studio One 2 Artist.license" will be saved to your computer. Copy this onto a piece of removable media and then transfer it onto the computer on which you want to run Studio One Artist.

4. Launch Studio One Artist.

   Go to the Activate Studio One Menu and click "Activate Offline."

5. Drag the "Studio One 2 Artist.license" file onto the Offline Activation window to complete the activation.

**Installing Bundled Content for Studio One Artist**


Install content from DVD or local folder
Use this if you purchased a boxed version of Studio One or if the content files are available in a local folder.

Studio One Artist comes bundled with more than 6 GB of plug-ins, loops, and samples, as well as an array of demo and tutorial material and other third-party content.

Once you've completed the Studio One Artist installation and activation process, the Studio One Content Installer will appear.

1. Click on "Install content from My PreSonus user account."

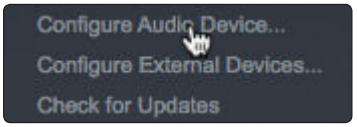
2. At the top of the installation menu, select the location where you wish to install the content. Listed in the installation menu are separate entries for each available item. Click the check box next to each item you wish to install, then click on the Install Packets button at the bottom left of the menu to install the selected content. When finished installing content, click on the Done button to exit the menu.

***Power User Tip:*** *If you choose not to install any portion the Studio One Artist content, you can install at any time in the future by going to the Studio One >Studio One Installation... menu.*

## 4.2 Enabling the Audio Driver

Studio One Artist was designed with PreSonus interfaces in mind, so setting up your AudioBox iOne or iTwo is quick and easy.

**Start Page: Setup Area**




**Selecting a Different Audio Driver from the Start Page**

If you don't see the correct driver for your interface ("PreSonus AudioBox iTwo" or "PreSonus AudioBox iOne") on the Start page when you launch Studio One, click the Configure Audio Devices link in the Setup area to open the Audio Setup Options window.

In the Audio Device menu, select the correct driver for your PreSonus AudioBox interface and then click the Apply and OK buttons.

After you have verified that the correct PreSonus AudioBox driver has been detected, please continue to the next section to set up your external MIDI devices. If you don't have any MIDI devices to connect at this time, ***please skip to Section 4.4***.

## 4.3 Configuring MIDI Devices (AudioBox iTwo)

From the External Devices window in Studio One Artist, you can configure your MIDI keyboard controller, sound modules, and control surfaces. This section will guide you through setting up your MIDI keyboard controller and sound module. Please consult the Reference Manual located on your Studio One Artist installation DVD for complete setup instructions for other MIDI devices.

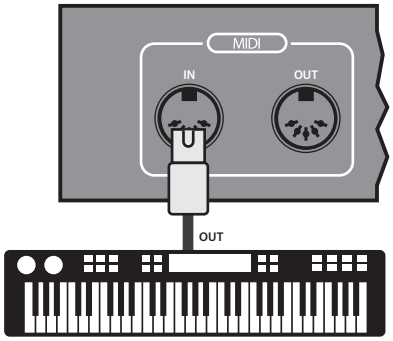
If you're using a third-party MIDI interface or USB MIDI-controller keyboard, you must install the drivers for these devices before proceeding with this section. Please consult the documentation that came with your MIDI hardware for complete installation instructions.

**Setting Up an External MIDI Keyboard Controller from the Start Page**

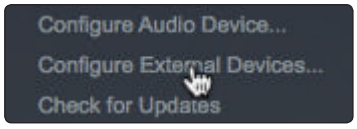
A MIDI keyboard controller is a hardware device that is generally used for playing and controlling other MIDI devices, virtual instruments, and software parameters. In Studio One Artist, these devices are referred to as Keyboards, and they must be configured before they are available for use. In some cases, your MIDI keyboard controller is also used as a tone generator. Studio One Artist views the controller and tone-generation functions as two different devices: a MIDI keyboard controller and a sound module. The MIDI controls (keyboard, knobs, faders, etc.) will be set up as a Keyboard. The sound module will be set up as an Instrument.

You can set up your external MIDI devices from the Setup area in the Start page. Before setting up a new Song for recording, take a moment to configure external devices.

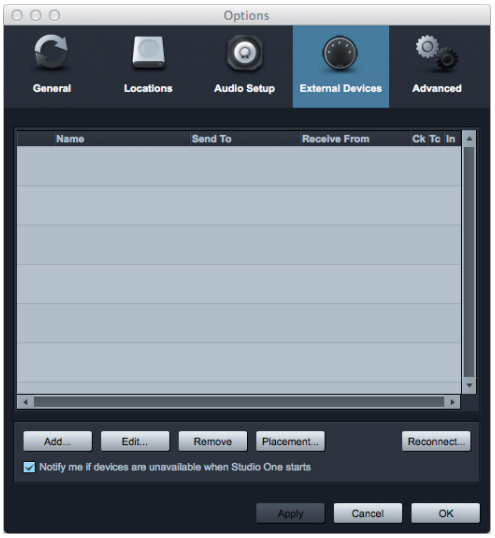
1. Connect the MIDI Out of your external MIDI controller to a MIDI In on your AudioBox iTwo or other MIDI interface. If you're using a USB MIDI controller, connect it to your computer and power it on.




2. Click on the Configure External Devices link in the Setup area on the Start page to launch the External Devices window.



3. Click the Add button. This opens the Add Device window.



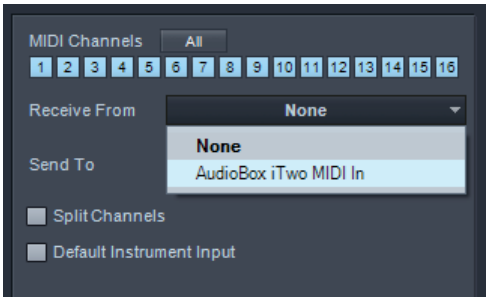

4. From the menu on the left, select New Keyboard. At this point, you can customize the name of your keyboard by entering the manufacturer and device names.

5. You must specify which MIDI channels will be used to communicate with this keyboard. For most purposes, you should select all MIDI channels. If you're unsure of which MIDI channels to choose, select all 16.

6. In the Receive From drop-down menu, select the MIDI-interface input from which Studio One Artist will receive MIDI data. In this case, it's the AudioBox iTwo.

7. In the Send To drop-down menu, select the MIDI interface output (AudioBox iTwo) from which Studio One Artist will send MIDI data to your keyboard.

***Power User Tip:*** *If your keyboard controller doesn't need to receive MIDI data from Studio One, you can leave this unselected. If your keyboard does need to receive MIDI data, you must connect a MIDI cable from the MIDI Out of the MIDI interface to the MIDI In of the keyboard.*




8. If this is the only keyboard you'll use for controlling your external synthesizers and virtual instruments, you should check the box next to Default Instrument Input. This automatically assigns your keyboard to control all MIDI devices in Studio One Artist.

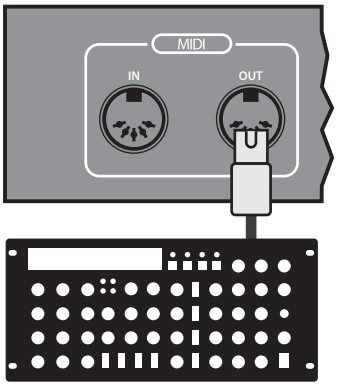
**Click OK.**

If you have a sound module that you'd like to connect, leave the External Devices window open and proceed to the next part of this section. If not, you can close this window and skip to ***Section 4.4***.

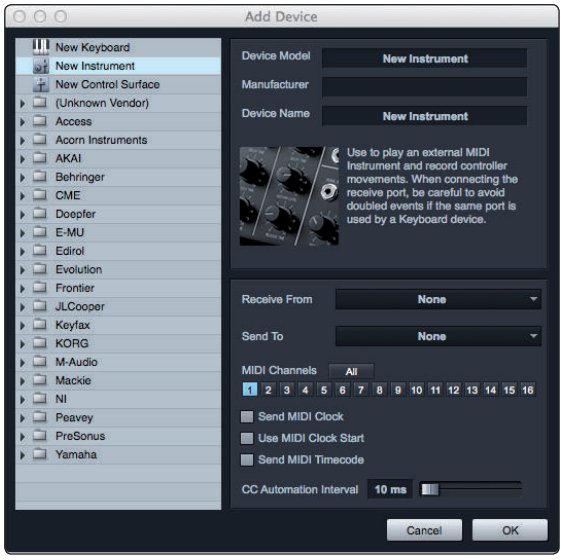
### Setting up an External MIDI Sound Module from the Start Page

MIDI instrument controllers (keyboards, MIDI guitars, etc.) send musical information in the form of MIDI data to sound modules and virtual instruments, which respond by generating sound, as instructed. Sound modules can be standalone devices or integrated into a MIDI instrument, such as a keyboard synthesizer. Studio One Artist refers to all sound generators as Instruments. Once you've set up your MIDI keyboard controller, take a moment to configure your sound module.

1. Connect the MIDI In of your external sound module to the MIDI Out of your AudioBox iTwo or other MIDI interface.




2. In the External Devices window, click the Add button. The Add Device window will open.

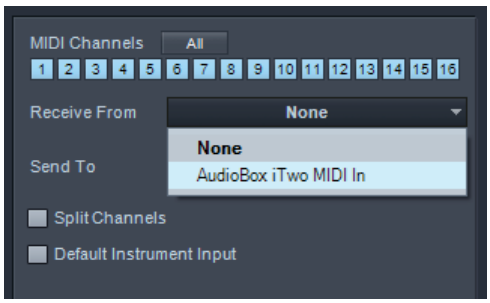



3. From the menu on the left, select "New Instrument." At this point, you can customize the name of your sound module by entering the manufacturer and device names.

4. Specify which MIDI channels will be used to communicate with this keyboard. For most purposes, you should select all MIDI channels. If you are unsure of which MIDI channels to select, we suggest you select all 16.

5. In the Send To drop-down menu, select the MIDI-interface output from which Studio One Artist will send MIDI data to your sound module—in this case, via the AudioBox iTwo.

6. In the Receive From drop-down menu, select the MIDI-interface output (AudioBox iTwo) from which Studio One Artist will receive sound-module MIDI data. If your sound module doesn't need to send information to Studio One, you can leave this unspecified.



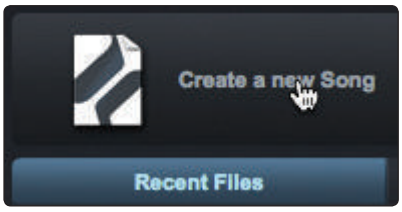

**Click OK** and close the External Devices window.

You're now ready to start recording in Studio One Artist.

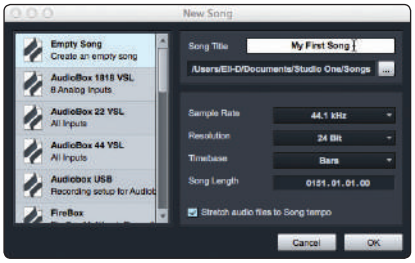
The rest of this Quick Start Guide will go over how to set up a Song and will discuss some general workflow tips for navigating through the Studio One Artist environment.

## 4.4 Creating a New Song

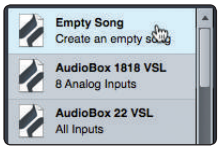
1. From the Start page, select "Create a new Song."



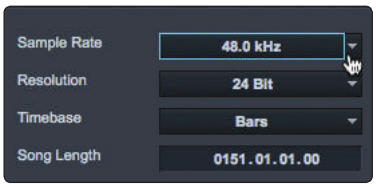

2. In the browser window, name your Song and choose the directory in which you'd like it saved.



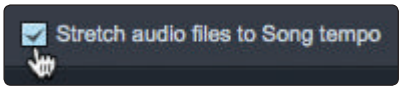

3. Select Empty Song from the Templates list. Note that you can select any one of the templates in the list to instantly create a pre-configured Session with tracks armed and ready for recording (similar to Capture). You can also create your own custom templates. Review the Studio One Reference Manual for more information.



4. Make sure the Sample Rate matches the rate set on your AudioBox iOne or iTwo. You can also determine the length of your Song and the type of time format you would like to use (Notation Bars, Seconds, Samples, or Frames).

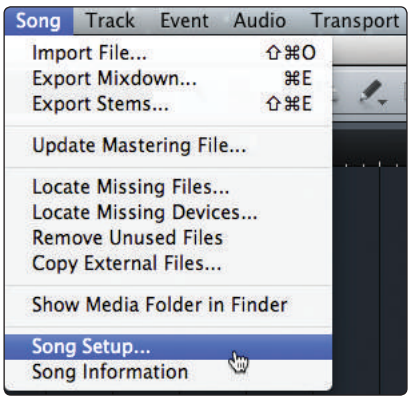



5. If you plan on importing loops into your Song, you may want to select Stretch Audio Loops to Song Tempo so that any loop of a known BPM (like those in the included content library) will import at the correct tempo.

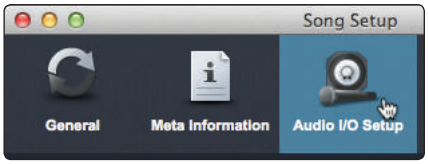


6. Click the OK button when you are finished.

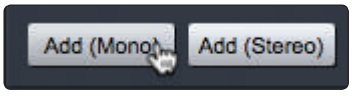
7. Go to Song>Song Setup menu.

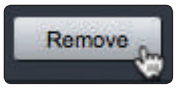



8. Click on "Audio I/O Setup."



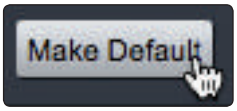

9. We recommend that you create a mono input for each of the inputs on your AudioBox iOne or iTwo. If you plan on recording in stereo, you should also create a stereo bus and assign it to the appropriate set of inputs.




10. You can remove any bus by simply selecting it and clicking the Remove button.

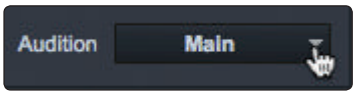
11. To customize the names of your buses, double-click on the default name to open a text box. When you have finished typing, hit Enter.

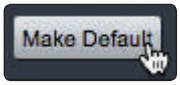



12. If you would like the same inputs to be available every time you launch Studio One Artist, click the "Make Default" button.

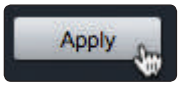
13. Click on the Outputs tab, and you will see all of the available outputs on your AudioBox iOne or iTwo.

14. In the lower right corner, you will see the Audition select menu. This allows you to choose the output from which you will audition audio files prior to importing them into Studio One Artist. In general, you will want this to be the main output bus.




15. If you would like this output configuration available every time you launch Studio One Artist, click the Make Default button.



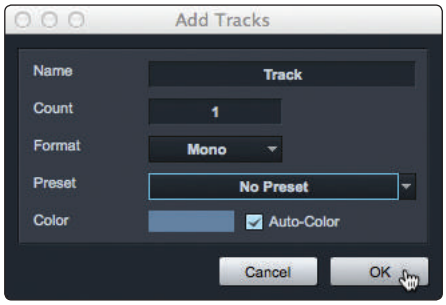

16. Click Apply to finish.

Now that you've configured your MIDI and audio I/O and created a new Song, let's go through some of the basics of Studio One Artist so you can start recording!

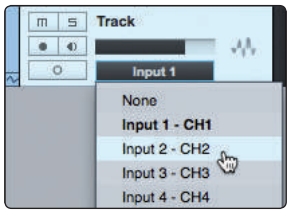
## 4.5 Creating Audio Tracks

In the upper left corner of the Arrange window, you will notice several buttons. The third button from the left is the Add Tracks button. Click this button to bring up the Add Tracks window.

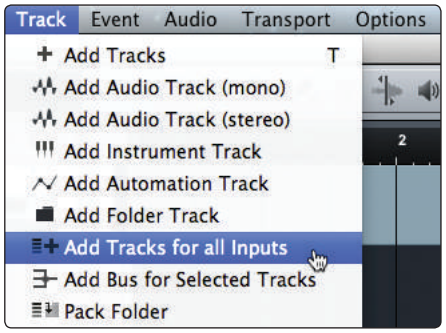
In the Add Tracks window, you can select the number and type of tracks you'd like to create (Mono Audio, Stereo Audio, Folder, Instrument, or Automation) and can customize the track name and color as well as add a preset rack of effects to the track.



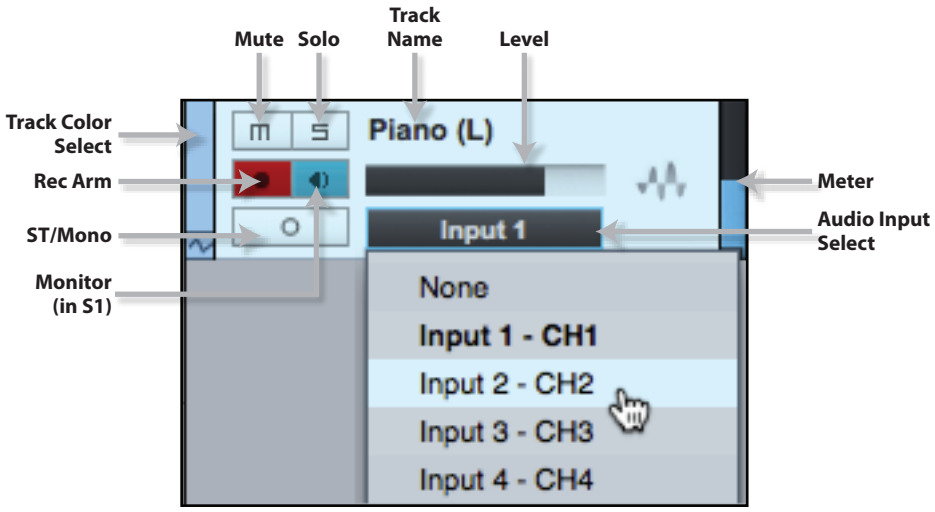

Once you have added your tracks, you can assign the input by simply clicking on the input to which a track is currently assigned. This will bring up the inputs list. You can also access the audio I/O setup from here.




If you would like to add a track for each of the available inputs and have the routing automatically assigned, simply go to Track | Add Tracks for All Inputs.



## 4.6 Anatomy of an Audio Track

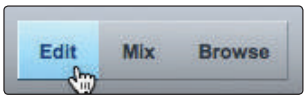



## 4.7 Adding Virtual Instruments and Plug-in Effects to Your Song

You can add plug-ins and instruments to your Song by dragging-and-dropping from the browser. You can also drag an effect or group of effects from one channel to another, drag in customized effects chains, and instantly load your favorite virtual-instrument patch without ever scrolling through a menu.

### Opening the Browser, Editor, and Mixer

In the lower right corner of the Arrange window are three buttons.

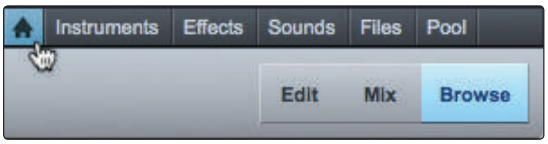



**The Edit button** opens or closes the audio editor or the MIDI piano-roll editor, depending on which type of track is selected.

**The Mix button** opens and closes the mixer window.

**The Browse button** opens the browser window, which displays all of the available virtual instruments, plug-in effects, audio files, and MIDI files, as well as the pool of audio files loaded into the current session.

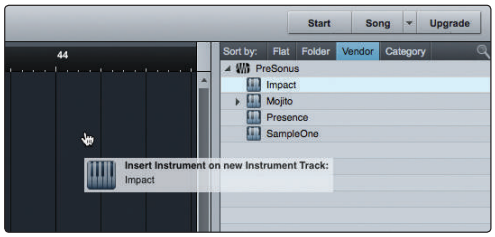
At the bottom of the browser window, you will find six category tabs:




- **Home** opens hot links to the other five tabs.
- **Instruments** opens a list of all the available Virtual Instruments that you have installed on your computer.
- **Effects** opens a list of all the available plug-in effects that you have installed on you computer.
- **Sounds** opens a list of the audio and MIDI loops that came with your Studio One content packages.
- **Files** provides you with quick access to every file on your computer.
- **Pool** opens a list of the audio files that have been recorded and imported into your current session.

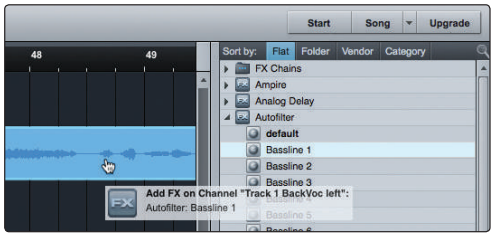
### 4.7.1 Drag-and-Drop Virtual Instruments

To add a virtual instrument to your session, click the Browse and Instrument buttons to open the instrument browser. Select the instrument or one of its patches from the instrument browser and drag it into the Arrange view. Studio One Artist will automatically create a new track and load the instrument as the input. For more information on Instrument and MIDI configuration, please review the Studio One Reference Manual.




### 4.7.2 Drag-and-Drop Effects

To add a plug-in effect to a track, click the Effects button and select the plug-in or one of its presets in the effects browser, then drag the selection over the track to which you would like to add the effect.



# 5.0 Tutorials

## 5.1 Microphone Types

The AudioBox iOne and iTwo work with most types of microphones, including dynamic, ribbon, and condenser microphones.

### 5.1.1 Condenser

Condenser microphones generally capture sound with excellent fidelity and are among the most popular microphone choices for studio recording and, increasingly, for live performance as well. Condenser microphones require a power source, which can be provided by a small battery, an external power supply, or phantom power, which is usually provided by a mixer, preamplifier, or direct (DI) box. Phantom power is sent over the same mic cable that carries the audio signal; the term derives from the fact that there is no visible power cord, and the voltage is not perceptible in the audio path. The AudioBox iOne and iTwo sends 48 VDC phantom power from the XLR inputs only.

### 5.1.2 Dynamic

Dynamic microphones are possibly the most widely used microphone type, especially in live shows. They are relatively inexpensive, resistant to physical damage, and typically handle high sound-pressure levels (SPL) very well. Unlike condenser microphones, most dynamic microphones do not require a power source.

Dynamic microphones, especially ribbon microphones, tend to generate low output voltages, so they typically need more preamp gain than condenser microphones.

**Ribbon**

Ribbon microphones are a special type of dynamic microphone and get their name from the thin metal ribbon used in their design. Ribbon microphones capture sound with very high fidelity—especially higher frequencies. However, they often are very fragile (many newer models are less so) and typically cannot handle high sound-pressure levels.

Most ribbon microphones do not require phantom power. In fact, unless a ribbon microphone specifically calls for phantom power, sending phantom power to a ribbon microphone can severely damage it—usually beyond repair.

### 5.1.3 USB Microphones and Other Types

Many microphone types are available, and as technology evolves, it is likely that more will be developed. One type of microphone to emerge recently is the USB microphone. These may be dynamic or condenser mics, but because a USB microphone is, in effect, an audio interface, we recommended that you not use them with the AudioBox iOne and iTwo, as the likelihood of conflicting drivers is great.

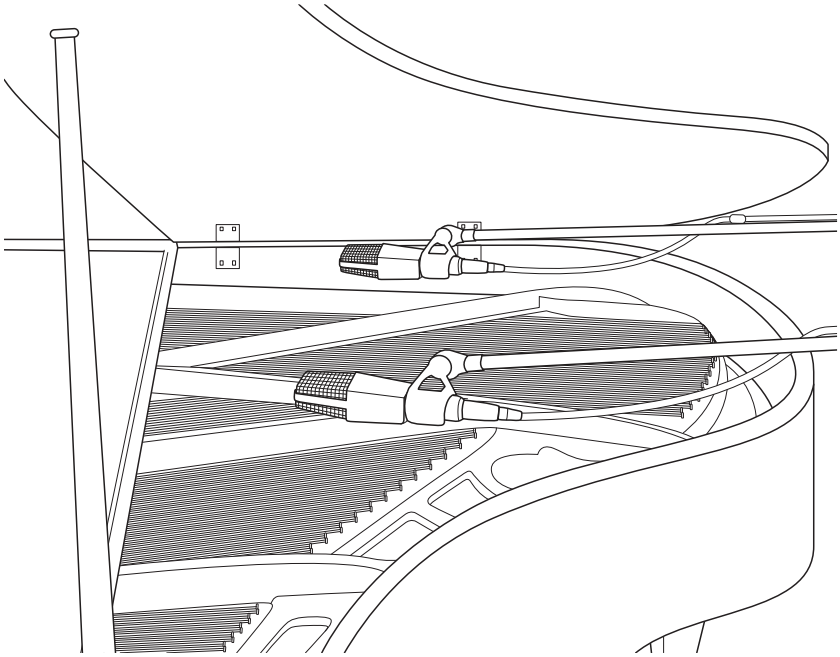
If you are using a new or nonstandard type of microphone (e.g., USB, headset, laser, MEMS), please consult your microphone's user's manual for power requirement and compatibility information.

Regardless of the microphone type you are using, we recommend reading your microphone's user's manual thoroughly before engaging phantom power and in case other usage questions arise.

### 5.1.4 Stereo Microphone Placement

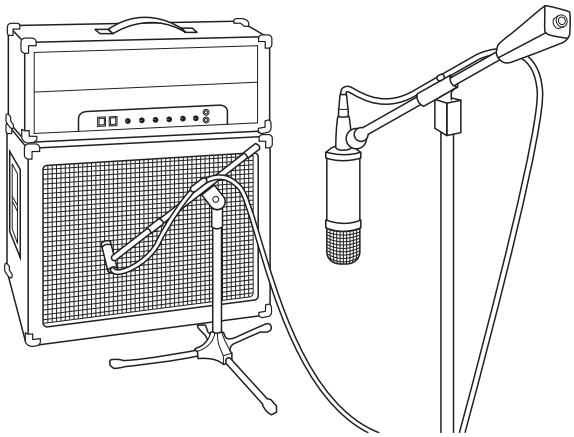
The following are a few recording applications to help you get started with your AudioBox iOne and iTwo. These are by no means the only ways to record these instruments. Microphone selection and placement is an art. For more information, visit your library or local bookstore, as there are many books and magazines about recording techniques. The Internet is also a great source of recording information, as are instructional videos. Some of these microphone-placement suggestions can be used in live applications, as well as for studio recording.

**Grand Piano**



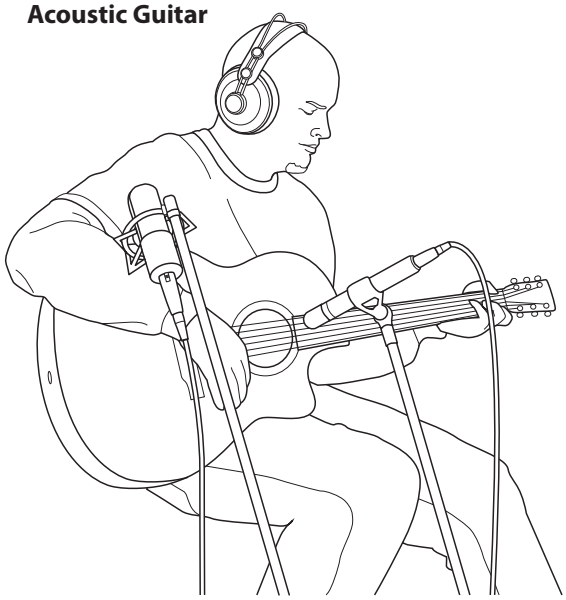
Place one microphone above the high strings and one microphone above the low strings. Experiment with distance (the farther back the more room you will capture). This technique can be used for live and studio applications.

**Electric Guitar**



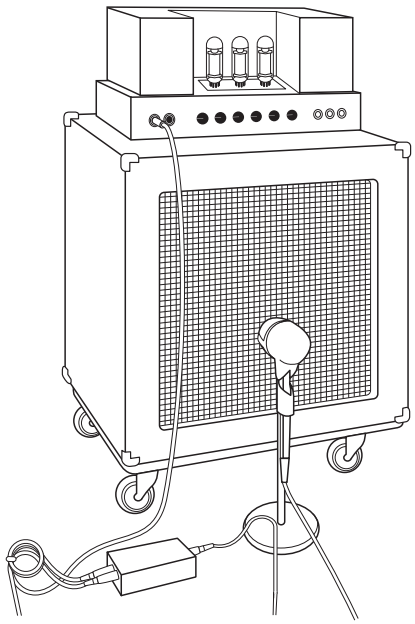
Place a dynamic microphone an inch or two away from the speaker of the guitar amplifier. Experiment with exact location. If you are recording an amp with multiple speakers, experiment with each one to see if one sounds better than the others. Place a condenser microphone approximately six feet away, pointed at the amp. Experiment with distance. Also experiment with inverting the phase of the room microphone to check for phase cancellation and reinforcement. (Select the "fuller"-sounding position.) To use this technique in a live application, omit the condenser microphone.

**Acoustic Guitar**



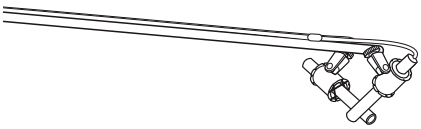
Point a small-diaphragm condenser microphone at the 12th fret, approximately 8 inches away. Point a large-diaphragm condenser microphone at the bridge of the guitar, approximately 12 inches from the guitar. Experiment with distances and microphone placement. Another popular method is using an XY microphone placement with two small-diaphragm condenser microphones. (See drum-overheads picture on the next page.)

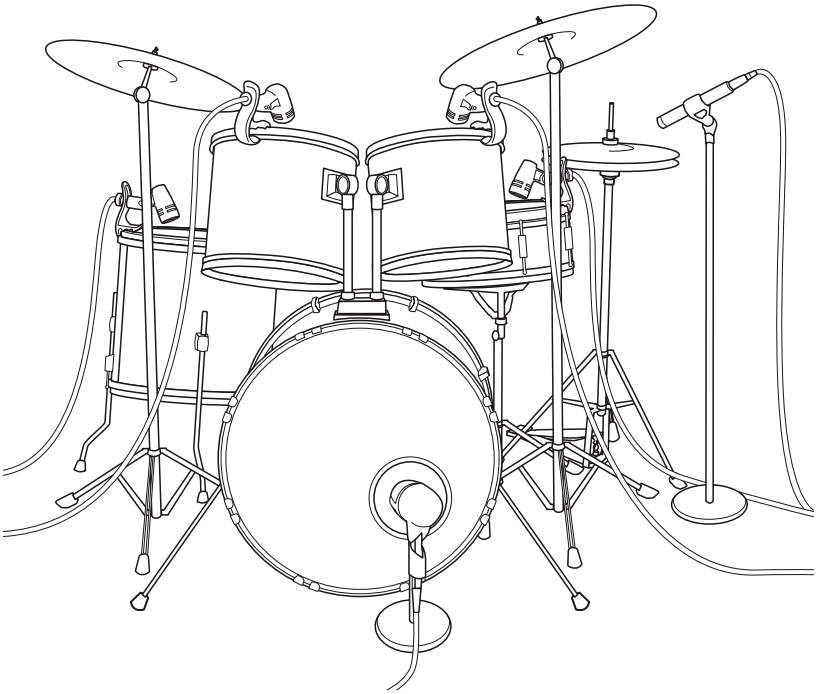
**Bass Guitar (Direct and Speaker)**



Plug the electric bass guitar into a passive direct box. Connect the instrument output from the passive direct box to a bass amplifier. Place a dynamic microphone an inch or two away from the speaker and connect it to an AudioBox microphone input. Connect the line output from the passive direct box to a line input on the AudioBox. For recording, place these signals on separate tracks. During mixing, you can blend the direct and amplifier signal to taste. This technique can also be used in live applications.

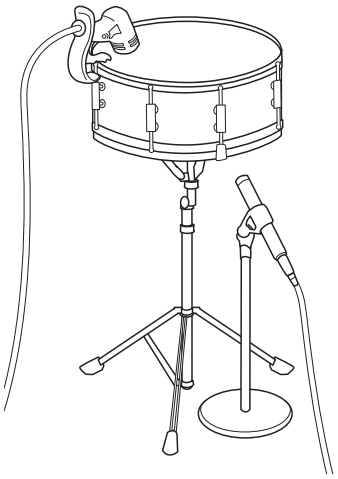
**Drum Overheads (XY example)**





Place two small-diaphragm condenser microphones on an XY stereo-microphone holder (bar). Position the microphones so that each one is at a 45-degree angle, pointed down at the drum kit, approximately 7 or 8 feet above the floor or drum riser. Experiment with height. This technique can be used in live applications as well.

**Snare Drum (top and bottom)**



Point a dynamic microphone at the center of the snare, making sure it is placed so that the drummer will not hit it. Place a small-diaphragm condenser microphone under the drum, pointed at the snares. Experiment with the placement of both microphones. Also experiment with inverting the phase of the bottom microphone. This technique can be used in live applications.

## 5.2 A Brief Tutorial on Dynamics Processing

Studio One Artist software is included with the AudioBox iOne and iTwo. This powerful DAW includes an assortment of dynamics plug-ins. What follows is an excerpt from a brief tutorial on dynamics processing written by PreSonus president and founder Jim Odom. It is included to help you get the most out of Studio One Artist. This tutorial will take you through the basics of dynamics processing and will explain the various types of dynamics processors.

### 5.2.1 Common Questions Regarding Dynamics Processing

**What is dynamic range?**

Dynamic range can be defined as the ratio between the loudest possible audio level and the lowest possible level. For example, if a processor states that the maximum input level before distortion is +24 dBu, and the output noise floor is -92 dBu, then the processor has a total dynamic range of 24 + 92 = 116 dB.

The average dynamic range of an orchestral performance can range from -50 dBu to +10 dBu, on average. This equates to a 60 dB dynamic range. Although 60 dB may not appear to be a large dynamic range, do the math, and you'll discover that +10 dBu is 1,000 times louder than -50 dBu!

Rock music, on the other hand, has a much smaller dynamic range: typically -10 dBu to +10 dBu, or 20 dB. This makes mixing the various signals of a rock performance together a much more tedious task.

**Why do we need compression?**

Consider the previous discussion: You are mixing a rock performance with an average dynamic range of 20 dB. You wish to add an uncompressed vocal to the mix. The average dynamic range of an uncompressed vocal is around 40 dB. In other words, a vocal performance can go from -30 dBu to +10 dBu. The passages that are +10 dBu and higher will be heard over the mix. However, the passages that are at -30 dBu and below will never be heard over the roar of the rest of the mix. A compressor can be used in this situation to reduce (compress) the dynamic range of the vocal to around 10 dB. The vocal can now be placed at around +5 dBu. At this level, the dynamic range of the vocal is from 0 dBu to +10 dBu. The lower level phrases will now be well above the lower level of the mix, and louder phrases will not overpower the mix, allowing the vocal to "sit in the track."

The same points can be made about any instrument in the mix. Each instrument has its place, and a good compressor can assist the engineer in the overall blend.

**Does every instrument need compression?**

This question may lead many folks to say "absolutely not, overcompression is horrible." That statement can be qualified by defining overcompression. The term itself must have been derived from the fact that you can hear the compressor working. A well-designed and properly adjusted compressor should not be audible! Therefore, the overcompressed sound is likely to be an improper adjustment on a particular instrument—unless, of course, it is done intentionally for effect.

**Why do the best consoles in the world put compressors on every channel?**

The answer is simply that most instruments need some form of compression, often very subtle, to be properly heard in a mix.

**Why do we need noise gates?**

Consider the compressed-vocal example discussed earlier; you now have a 20 dB dynamic range for the vocal channel. Problems arise when noise or instruments (air conditioner, loud drummer, etc.) in the background of the vocal mic become more audible after the lower end of the dynamic range is raised. You might attempt to mute the vocal between phrases in an attempt to remove the unwanted sounds; however, this would probably end disastrously. A better method is to use a noise gate. The noise-gate threshold could be set at the bottom of the dynamic range of the vocal, say -10 dBu, such that the gate would shut out the unwanted signals between the phrases.

If you have ever mixed live sound, you know the problems cymbals can create by bleeding through the tom mics. As soon as you add some highs to get some snap out of the tom, the cymbals come crashing through, placing the horn drivers into a small orbit. Gating those tom mics so that the cymbals no longer ring through them will give you an enormous boost in cleaning up the overall mix.

Dynamics processing is the process of altering the dynamic range of a signal, thereby enhancing the ability of a live sound system or recording device to handle the signal without distortion or noise and aiding in placing the signal in the overall mix.

### 5.2.2 Types of Dynamic Processing

**Compression/Limiting**

Punch, apparent loudness, presence—these are just three of the many terms used to describe the effects of compression/limiting.

Compression and limiting are forms of dynamic-range (gain) control. Audio signals have very wide peak-to-average signal-level ratios (sometimes referred to as dynamic range, which is the difference between the loudest level and the softest level). The peak signal can cause overload in the audio-recording or sound-reinforcement chain, resulting in signal distortion.

A compressor/limiter is a type of amplifier in which gain is dependent on the signal level passing through it. You can set the maximum level a compressor/limiter allows to pass through, thereby causing automatic gain reduction above some predetermined signal level, or threshold. Compression refers, basically, to the ability to reduce, by a fixed ratio, the amount by which a signal's output level can increase relative to the input level. It is useful for lowering the dynamic range of an instrument or vocal, making it easier to record without distorting the recorder. It also assists in the mixing process by reducing the amount of level changes needed for a particular instrument.

Take, for example, a vocalist who moves around in front of the microphone while performing, making the output level vary up and down unnaturally. A compressor can be applied to the signal to help correct this recording problem by reducing the louder passages enough to be compatible with the overall performance.

How severely the compressor reduces the signal is determined by the compression ratio and compression threshold. A ratio of 2:1 or less is considered mild compression, reducing the output by a factor of two for signals that exceed the compression threshold. Ratios above 10:1 are considered hard limiting.

As the compression threshold is lowered, more of the input signal is compressed (assuming a nominal input-signal level). Care must be taken not to overcompress a signal, as too much compression destroys the acoustic dynamic response of a performance. (That said, overcompression is used by some engineers as an effect, with killer results!)

Limiting refers to the processing that prevents the signal from getting any louder (that is, it prevents any increase in the signal's amplitude) at the output.

Compressor/limiters are commonly used for many audio applications. For example:

A kick drum can get lost in a wall of electric guitars. No matter how much the level is increased, the kick drum stays lost in the "mud." A touch of compression can tighten up that kick-drum sound, allowing it to punch through without having to crank the level way up.

A vocal performance usually has a wide dynamic range. Transients (normally the loudest portions of the signal) can be far outside the average level of the vocal signal. Because the level can change continuously and dramatically, it is extremely difficult to ride the level with a console fader. A compressor/limiter automatically controls gain without altering the subtleties of the performance.

A solo guitar can seem to be masked by the rhythm guitars. Compression can make your lead soar above the track without shoving the fader through the roof.

Bass guitar can be difficult to record. A consistent level with good attack can be achieved with proper compression. Your bass doesn't have to be washed out in the low end of the mix. Let the compressor/limiter give your bass the punch it needs to drive the bottom of the mix.

**Compressors — Terminology**

**Threshold.** The compressor threshold sets the level at which compression begins. When the signal is above the threshold setting, it becomes eligible for compression. Basically, as you turn the threshold knob counterclockwise, more of the input signal becomes compressed (assuming you have a ratio setting greater than 1:1).

**Ratio**. The ratio is the relationship between the output level and the input level. In other words, the ratio sets the compression slope. For example, if you have the ratio set to 2:1, any signal levels above the threshold setting will be compressed such that for every 1 dB of level increase into the compressor, the output will only increase 0.5 dB. As you increase the ratio, the compressor gradually becomes a limiter.

**Limiter**. A limiter is a compressor that is set to prevent any increase in the level of a signal above the threshold. For example, if you have the threshold knob set at 0 dB, and the ratio turned fully clockwise, the compressor becomes a limiter at 0 dB, so that the output signal cannot exceed 0 dB regardless of the level of the input signal.

**Attack**. Attack sets the speed at which the compressor acts on the input signal. A slow attack time allows the beginning envelope of a signal (commonly referred to as the initial transient) to pass through the compressor unprocessed, whereas a fast attack time immediately subjects the signal to the ratio and threshold settings of the compressor.

**Release**. Release sets the length of time the compressor takes to return the gain reduction back to zero (no gain reduction) after the signal level drops below the compression threshold. Very short release times can produce a very choppy or "jittery" sound, especially in low-frequency instruments such as bass guitar. Very long release times can result in an overcompressed sound; this is sometimes referred to as "squashing" the sound. All ranges of release can be useful at different times, however, and you should experiment to become familiar with the different sonic possibilities.

**Hard/Soft Knee**. With hard-knee compression, the gain reduction applied to the signal occurs as soon as the signal exceeds the level set by the threshold. With soft-knee compression, the onset of gain reduction occurs gradually after the signal has exceeded the threshold, producing a more musical response (to some folks).

**Auto**. Places a compressor in automatic attack and release mode. The attack and release knobs become inoperative and a preprogrammed attack and release curve is used.

**Makeup Gain**. When compressing a signal, gain reduction usually results in an overall reduction of level. The gain control allows you to restore the loss in level due to compression (like readjusting the volume).

**Compressor Sidechain**. The sidechain jack interrupts the signal that the compressor is using to determine the amount of gain reduction it should apply. When no connector is inserted into this jack, the input signal goes directly to the compressor's control circuitry. When a connector is inserted into this jack, the signal path is broken. The control signal can then be processed by an equalizer, for example, to reduce sibilance (de-essing) in a vocal track. The control signal is then returned to the unit via the connector. One common application for a sidechain is when using a compressor to reduce the level of music or other background sound whenever a narrator speaks or vocalist sings, allowing the voice to be clearly heard. In this application, the vocal signal is routed to the sidechain input, while the music is routed through the main compression circuitry. Now the compressor will automatically duck—that is, reduce the level of—the music whenever the narrator speaks or the vocalist sings.

### Expansion

There are two basic types of expansion: dynamic and downward. Expansion increases the dynamic range of a signal after the signal crosses the expansion threshold. Dynamic expansion is basically the opposite of compression. In fact, broadcasters use dynamic expansion to "undo" compression before transmitting the audio signal. This is commonly referred to as *companding,* or COMPression followed by expANDING.

By far the most common use of expansion is downward expansion. In contrast to compression, which decreases the level of a signal after it rises above the compression threshold, expansion decreases the level of a signal after the signal goes below the expansion threshold. The amount of level reduction is determined by the expansion ratio. For example, a 2:1 expansion ratio reduces the level of a signal by a factor of two. (e.g., if a level drops 5 dB below the expansion threshold, the expander will reduce it to 10 dB below the threshold.)

Commonly used for noise reduction, expansion is very effective as a simple noise gate. The major difference between expansion and noise gating is that expansion is dependent on the signal level after the level crosses the threshold, whereas a noise gate works independent of a signal's level beyond the threshold.

### Expansion— Terminology

**Downward Expansion**. Downward expansion is the most common expansion used in live sound and recording. This type of expansion reduces the level of a signal when the signal falls below a set threshold level. This is most common used for noise reduction.

**Ratio**. The expansion ratio sets the amount of reduction applied to a signal once the signal has dropped below the expansion threshold. For example, a 2:1 expansion ratio attenuates a signal 2 dB for every 1 dB it drops below the threshold. Ratios of 4:1 and higher act much like a noise gate but without the ability to tailor the attack, hold, and release times.

### Noise Gates

**Threshold**. The gate threshold sets the level at which the gate opens. Essentially, all signals above the threshold setting are passed through unaffected, whereas signals below the threshold setting are reduced in level by the amount set by the range control. If the threshold is set fully counterclockwise, the gate is turned off (always open), allowing all signals to pass through unaffected.

**Attack**. The gate attack time sets the rate at which the gate opens. A fast attack rate is crucial for percussive instruments, whereas signals such as vocals and bass guitar require a slower attack. Too fast of an attack can, on these slow-rising signals, cause an artifact in the signal, which is heard as a click. All gates have the ability to click when opening but a properly set gate will never click.

**Hold**. Hold time is used to keep the gate open for a fixed period after the signal drops below the gate threshold. This can be very useful for effects such as gated snare, where the gate remains open after the snare hit for the duration of the hold time, then abruptly closes.

**Release**. The gate-release time determines the rate at which the gate closes. Release times should typically be set so that the natural decay of the instrument or vocal being gated is not affected. Shorter release times help to clean up the noise in a signal but may cause "chattering" in percussive instruments. Longer release times usually eliminate "chattering" and should be set by listening carefully for the most natural release of the signal.

**Range**. The gate range is the amount of gain reduction that the gate produces. Therefore, if the range is set at 0 dB, there will be no change in the signal as it crosses the threshold. If the range is set to -60 dB, the signal will be gated (reduced) by 60 dB, etc.

**Key Listen**. The key listen allows the user to listen to the signal that is being filtered by the gate.

**Frequency Key Filter**. Some gates offer a variable frequency control allowing the user to set a specific frequency band that the will cause the gate to open or close.

**Noise Gating.** Noise gating is the process of removing unwanted sounds from a signal by attenuating all signals below a set threshold. As described, the gate works independently of the audio signal after being "triggered" by the signal crossing the gate threshold. The gate will remain open as long as the signal is above the threshold. How fast the gate opens to let the "good" signal through is determined by the attack time. How long the gate stays open after the signal has gone below the threshold is determined by the hold time. How fast the gate closes is determined by the release. How much the gate attenuates the unwanted signal while closed is determined by the range.

Noise gates were originally designed to help eliminate extraneous noise and unwanted artifacts from a recording, such as hiss, rumble, or transients from other instruments in the room. Since hiss and noise are not as loud as the instrument being recorded, a properly set gate will only allow the intended sound to pass through; the volume of everything else is lowered. Not only will this strip away unwanted artifacts like hiss, it will add definition and clarity to the desired sound. This is a very popular application for noise gates, especially with percussion instruments, as it will add punch or "tighten" the percussive sound and make it more pronounced.

## 5.2.3 Compression Setting Suggestions

The following are the compression presets that were used in the PreSonus BlueMax. We have included them as a jumping-off point for setting up compression in Studio One Artist.

### Vocals

**Soft.** This is an easy compression with a low ratio setting for ballads, allowing a wider dynamic range. It's good for live use. This setting helps the vocal "sit in the track."

| THRESHOLD | RATIO | ATTACK | RELEASE |
|---|---|---|---|
| -8.2 dB | 1.8:1 | 0.002 ms | 38 ms |

**Medium.** This setting has more limiting than the Soft compression setting, producing a narrower dynamic range. It moves the vocal more up front in the mix.

| THRESHOLD | RATIO | ATTACK | RELEASE |
|---|---|---|---|
| -3.3 dB | 2.8:1 | 0.002 ms | 38 ms |

**Screamer.** This setting is for loud vocals. It is a fairly hard compression setting for a vocalist who is on and off the microphone a lot. It puts the voice "in your face."

| THRESHOLD | RATIO | ATTACK | RELEASE |
|---|---|---|---|
| -1.1 dB | 3.8:1 | 0.002 ms | 38 ms |

### Percussion

**Snare/Kick.** This setting allows the first transient through and compresses the rest of the signal, giving a hard "snap" up front and a longer release.

| THRESHOLD | RATIO | ATTACK | RELEASE |
|---|---|---|---|
| -2.1 dB | 3.5:1 | 78 ms | 300 ms |

**Left/Right (Stereo) Overheads.** The low ratio and threshold in this setting gives a "fat" contour to even out the sound from overhead drum mics. Low end is increased, and the overall sound is more present and less ambient. You get more "boom" and less "room."

| THRESHOLD | RATIO | ATTACK | RELEASE |
|---|---|---|---|
| -13.7 dB | 1.3:1 | 27 ms | 128 ms |

### Fretted Instruments

**Electric Bass.** The fast attack and slow release in this setting will tighten up the electric bass and give you control for a more consistent level.

| THRESHOLD | RATIO | ATTACK | RELEASE |
|---|---|---|---|
| -4.4 dB | 2.6:1 | 45.7 ms | 189 ms |

**Acoustic Guitar.** This setting accentuates the attack of the acoustic guitar and helps maintain an even signal level, keeping the acoustic guitar from disappearing in the track.

| THRESHOLD | RATIO | ATTACK | RELEASE |
|---|---|---|---|
| -6.3 dB | 3.4:1 | 188 ms | 400 ms |

**Electric Guitar.** This is a setting for "crunch" electric rhythm guitar. A slow attack helps to get the electric rhythm guitar "up close and personal" and gives punch to your crunch.

| THRESHOLD | RATIO | ATTACK | RELEASE |
|---|---|---|---|
| -0.1 dB | 2.4:1 | 26 ms | 193 ms |

### Keyboards

**Piano.** This is a special setting for an even level across the keyboard. It is designed to help even up the top and bottom of an acoustic piano. In other words, it helps the left hand to be heard along with the right hand.

| THRESHOLD | RATIO | ATTACK | RELEASE |
|---|---|---|---|
| -10.8 dB | 1.9:1 | 108 ms | 112 ms |

**Synth.** The fast attack and release on this setting can be used for synthesizer horn stabs or for bass lines played on a synthesizer.

| THRESHOLD | RATIO | ATTACK | RELEASE |
|---|---|---|---|
| -11.9 dB | 1.8:1 | 0.002 ms | 85 ms |

**Orchestral.** Use this setting for string pads and other types of synthesized orchestra parts. It will decrease the overall dynamic range for easier placement in the mix.

| THRESHOLD | RATIO | ATTACK | RELEASE |
|---|---|---|---|
| 3.3 dB | 2.5:1 | 1.8 ms | 50 ms |

### Stereo Mix

**Stereo Limiter.** Just as the name implies, this is a hard limiter, or "brickwall," setting—ideal for controlling the level to a two-track mixdown deck or stereo output.

| THRESHOLD | RATIO | ATTACK | RELEASE |
|---|---|---|---|
| 5.5 dB | 7.1:1 | 0.001 ms | 98 ms |

**Contour.** This setting fattens up the main mix.

| THRESHOLD | RATIO | ATTACK | RELEASE |
|---|---|---|---|
| -13.4 dB | 1.2:1 | 0.002 ms | 182 ms |

### Effects

**Squeeze.** This is dynamic compression for solo work, especially electric guitar. It gives you that glassy "Tele/Strat" sound. It is a true classic.

| THRESHOLD | RATIO | ATTACK | RELEASE |
|---|---|---|---|
| -4.6 dB | 2.4:1 | 7.2 ms | 93 ms |

**Pump.** This is a setting for making the compressor "pump" in a desirable way. This effect is good for snare drums to increase the length of the transient by bringing the signal up after the initial spike.

| THRESHOLD | RATIO | ATTACK | RELEASE |
|---|---|---|---|
| 0 dB | 1.9:1 | 1 ms | 0.001 ms |

## 5.3 Equalizers

Studio One Artist also includes several EQ plug-ins. Here's a brief explanation of how an EQ functions, as well as some charts to help you navigate the frequency ranges of various instruments so you can quickly choose the best EQ settings for your recordings and live mixes.

### 5.3.1 What is an EQ?

An equalizer is a filter that allows you to adjust the level of a frequency, or range of frequencies, of an audio signal. In its simplest form, an EQ will let you turn the treble and bass up or down, allowing you to adjust the coloration of, let's say, your car stereo or iPod. In recording, equalization is a sophisticated art. Good equalization is critical to a good mix.

When used correctly, an equalizer can provide the impression of nearness or distance, "fatten" or "thin" a sound, and help blend or provide separation between similar sounds in a mix allowing them to both shine through the mix.

**Parametric EQ**

The parametric EQ and semi-parametric EQ are the most common equalizers found in recording and live situations because they offer continuous control over all parameters. A parametric EQ offers continuous control over the audio signal's frequency content, which is divided into several bands of frequencies (most commonly three to seven bands). A fully parametric EQ like those in the StudioLive 24.4.2 offers control over the bandwidth (basically, the range of frequencies affected), the center frequency of the band, and the level (boost/cut) of the designated frequency band. It also offers separate control over the Q, which is the ratio of the center frequency to the bandwidth. A semi-parametric EQ provides control over most of these parameters but the Q is fixed. Some devices, such as the StudioLive 16.4.2 and 16.0.2 and the AudioBox VSL-series interfaces, have quasi-parametric EQ, which is semi-parametric EQ with a simple, switchable Q setting (typically, High and Low Q).

**Q**

Q is the ratio of center frequency to bandwidth, and if the center frequency is fixed, then bandwidth is inversely proportional to Q—meaning that as you raise the Q, you narrow the bandwidth. In fully parametric EQs, you have continuous bandwidth control and/or continuous Q control, which allows you to attenuate or boost a very narrow or wide range of frequencies.

A narrow bandwidth (higher Q) has obvious benefits for removing unpleasant tones. Let's say the snare drum in your mix has an annoying ring to it. With a very narrow bandwidth, you can isolate this one frequency (usually around 1 kHz) and remove, or reject, it. This type of narrow band-reject filter is also known as a *notch filter*. By notching out the offending frequency, you can remove the problem without removing the instrument from the mix. A narrow bandwidth is also useful in boosting pleasant tones of an instrument such as the attack. Take for instance, a kick drum. A kick drum resonates somewhere between 60 to 125 Hz, but the attack of the kick drum is much higher at 2 to 5 kHz. By setting a narrow bandwidth and boosting the attack a bit, you can achieve a punchier kick drum without overpowering the rest of the mix.

A broad bandwidth accentuates or attenuates a larger band of frequencies. The broad and narrow bandwidths (high and low Q) are usually used in conjunction with one another to achieve the desired effect. Let's look at our kick drum again. We have a kick drum that has a great, big, low-end sound centered around 100 Hz and an attack hitting almost dead-on at 4 kHz. In this example, you would use a broad bandwidth in the low frequency band, centered at 100 Hz, and a narrow bandwidth boosted at 4 kHz. In this way you are accentuating the best and downplaying everything else this particular kick drum has to offer.

### Shelving EQ

A shelving EQ attenuates or boost frequencies above or below a specified cutoff point. Shelving equalizers come in two different varieties: high-pass and low-pass.

Low-pass shelving filters pass all frequencies below the specified cutoff frequency while attenuating all the frequencies above it. A high-pass filter does the opposite: passing all frequencies above the specified cut-off frequency while attenuating everything below.

### Graphic EQ

A graphic EQ is a multiband equalizer that uses sliders to adjust the amplitude for each frequency band. It gets its name from the positions of the sliders, which graphically display the resulting frequency-response curve. The center frequency and bandwidth are fixed; the level (amplitude) for each band is the only adjustable parameter.

Graphic EQs are generally used to fine-tune the overall mix for a particular room. For instance, if you are mixing in a "dead" room, you may want to boost high frequencies and roll off some of the lows. If you are mixing in a "live" room, you might need to lower the high-midrange and highest frequencies. In general, you should not make drastic amplitude adjustments to any particular frequency bands. Instead, make smaller, incremental adjustments over a wider spectrum to round out your final mix. To assist you with these adjustments, here is an overview of which frequencies affect different sound characteristics:

**Sub-Bass (16 Hz to 60 Hz).** The lowest of these bass frequencies are felt, rather than heard, as with freeway rumbling or an earthquake. These frequencies give your mix a sense of power even when they only occur occasionally. However, overemphasizing frequencies in this range will result in a muddy mix.

**Bass (60 Hz to 250 Hz).** Because this range contains the fundamental notes of the rhythm section, any EQ changes will affect the balance of your mix, making it fat or thin. Too much emphasis will make for a boomy mix.

**Low Mids (250 Hz to 2 kHz).** In general, you will want to emphasize the lower portion of this range and deemphasize the upper portion. Boosting the range from 250 Hz to 500 Hz will accent ambience in the studio and will add clarity to bass and lower frequency instruments. The range between 500 Hz and 2 kHz can make midrange instruments (guitar, snare, saxophone, etc.) "honky," and too much boost between 1 kHz and 2 kHz can make your mix sound thin or "tinny."

**High Mids (2 kHz to 4 kHz).** The attack portion of percussive and rhythm instruments occurs in this range. High mids are also responsible for the projection of midrange instruments.

**Presence (4 kHz to 6 kHz).** This frequency range is partly responsible for the clarity of a mix and provides a measure of control over the perception of distance. If you boost this frequency range, the mix will be perceived as closer to the listener. Attenuating around 5 kHz will make the mix sound further away but also more transparent.

**Brilliance (6 kHz to 16 kHz).** While this range controls the brilliance and clarity of your mix, boosting it too much can cause some clipping so keep an eye on your main meter.

### 5.3.2 Equalization Settings: How to Find the Best and Leave the Rest

How do you find the best and worst each instrument has to offer and adjust their frequency content accordingly? Here's a quick guide:

- First, solo just the instrument with which you are working. Most engineers start building their mix with the drums and work from the bottom up (kick, snare, toms, hi-hat, overheads). Each instrument resonates primarily in a specific frequency band, so if you are working on your kick-drum mic, start with the lowest band of the EQ. Tune in the best-sounding low end and move on to the attack. It is not uncommon to hear an annoying ringing or a "twang" mixed in with your amazing-sounding low end and perfect attack, so your next task will be to find that offending frequency and notch it out. Once you are satisfied with your kick drum, mute it, and move on to the next instrument.
- Taking your time with equalization is well worth the effort. Your mix will have better separation and more clarity.

**Additional advice:**

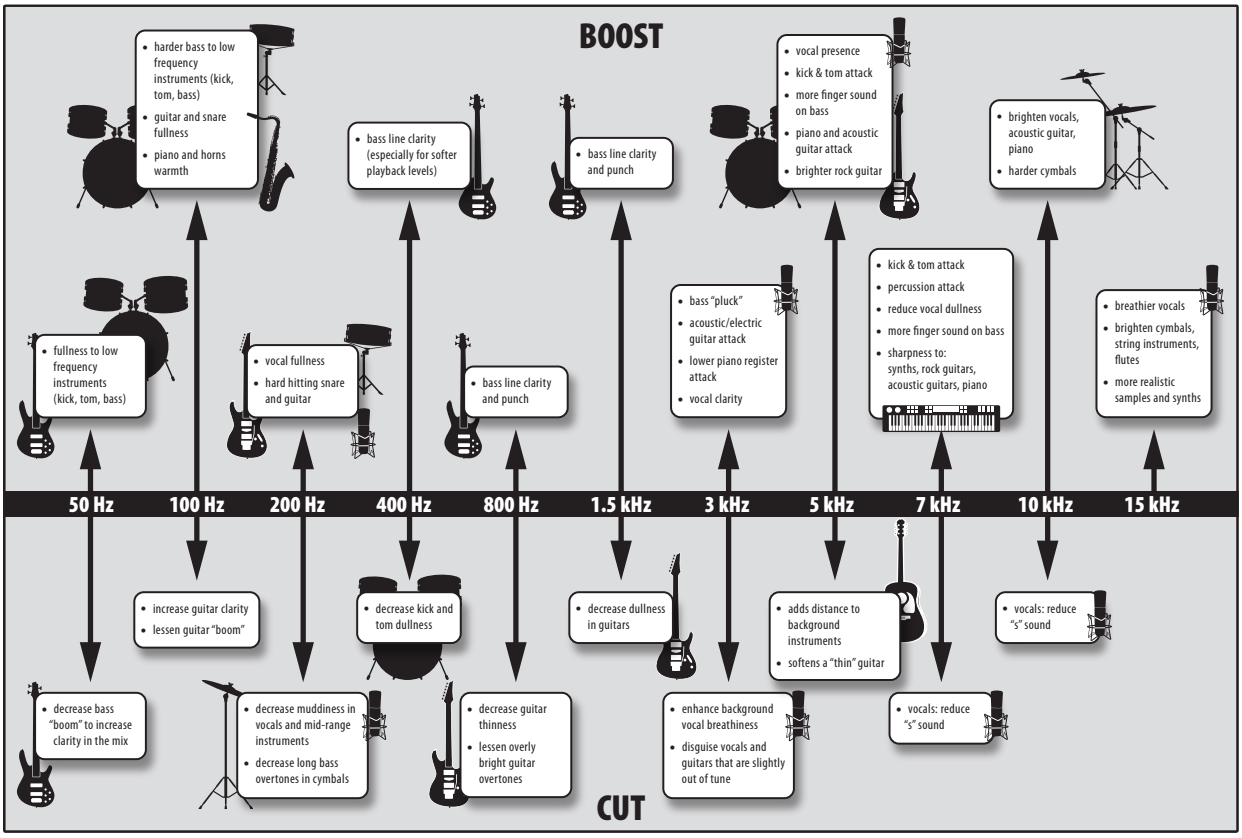
- **You can only do so much.** Not every instrument can or should have a full, rich low end and a sharp attack. If every instrument is EQ'd to have the same effect, it will lose its identity in the mix. Your goal is not individual perfection, it is perfection in unity.
- **Step away from the mix.** Your ears get fatigued, just like the rest of you. If you are working particularly hard on one instrument, your ears will be quite literally numbed to that frequency range.
- **Your memory is not what you think it is.** Comparing a flat EQ and the curve that you've created allows you to see and hear exactly what you've done. So be honest with yourself. Sometimes that EQ setting you've been working on for 15 minutes is not the right choice, so move on.
- **Never be afraid of taking a risk.** The best EQ tricks were found by mad scientists of sound. With every instrument, there are frequencies that can be attenuated or boosted to add clarity or fullness. Altering the wrong frequencies can make an instrument shrill, muddy, or just downright annoying. The following two charts suggest frequency ranges that should be accentuated or downplayed for the most common instruments. These are just suggestions; the frequencies may need to be adjusted up or down depending on the instrument, room, and microphone.

## 5.4 EQ Frequency Guides

**Table 1**

| Instrument | What to Cut | Why to Cut | What to Boost | Why to Boost |
|---|---|---|---|---|
| Human Voice | 7 kHz | Sibilance | 8 kHz | Big sound |
| | 2 kHz | Shrill | 3 kHz and above | Clarity |
| | 1 kHz | Nasal | 200-400 Hz | Body |
| | 80 Hz and below | Popping P's | | |
| Piano | 1-2 kHz | Tinny | 5 kHz | More presence |
| | 300 Hz | Boomy | 100 Hz | Bottom end |
| Electric Guitar | 1-2 kHz | Shrill | 3 kHz | Clarity |
| | 80 Hz and below | Muddy | 125 Hz | Bottom end |
| Acoustic Guitar | 2-3 kHz | Tinny | 5 kHz and above | Sparkle |
| | 200 Hz | Boomy | 125 Hz | Full |
| Electric Bass | 1 kHz | Thin | 600 Hz | Growl |
| | 125 Hz | Boomy | 80 Hz and below | Bottom end |
| String Bass | 600 Hz | Hollow | 2-5 kHz | Sharp attack |
| | 200 Hz | Boomy | 125 Hz and below | Bottom end |
| Snare Drum | 1 kHz | Annoying | 2 kHz | Crisp |
| | | | 150-200 Hz | Full |
| | | | 80 Hz | Deep |
| Kick Drum | 400 Hz | Muddy | 2-5 kHz | Sharp attack |
| | 80 Hz and below | Boomy | 60-125 Hz | Bottom end |
| Toms | 300 Hz | Boomy | 2-5 kHz | Sharp attack |
| | | | 80-200 Hz | Bottom end |
| Cymbals | 1 kHz | Annoying | 7-8 kHz | Sizzle |
| | | | 8-12 kHz | Brilliance |
| | | | 15 kHz | Air |
| Horns | 1 kHz | Honky | 8-12 kHz | Big sound |
| | 120 Hz and below | Muddy | 2 kHz | Clarity |
| String section | 3 kHz | Shrill | 2 kHz | Clarity |
| | 120 Hz and below | Muddy | 400-600 Hz | Lush and full |

**Table 2**


BOOST
• harder bass to low frequency instruments (kick, tom, bass)
• guitar and snare fullness
• piano and horns warmth
• bass line clarity (especially for softer playback levels)
• bass line clarity and punch
• vocal presence
• kick & tom attack
• more finger sound on bass
• piano and acoustic guitar attack
• brighter rock guitar
• brighten vocals, acoustic guitar, piano
• harder cymbals
• fullness to low frequency instruments (kick, tom, bass)
• vocal fullness
• hard hitting snare and guitar
• bass line clarity and punch
• bass "pluck"
• acoustic/electric guitar attack
• lower piano register attack
• vocal clarity
• kick & tom attack
• percussion attack
• reduce vocal dullness
• more finger sound on bass
• sharpness to: synths, rock guitars, acoustic guitars, piano
• breathier vocals
• brighten cymbals, string instruments, flutes
• more realistic samples and synths
50 Hz
100 Hz
200 Hz
400 Hz
800 Hz
1.5 kHz
3 kHz
5 kHz
7 kHz
10 kHz
15 kHz
• increase guitar clarity
• lessen guitar "boom"
• decrease kick and tom dullness
• decrease dullness in guitars
• adds distance to background instruments
• softens a "thin" guitar
• vocals: reduce "s" sound
• decrease bass "boom" to increase clarity in the mix
• decrease muddiness in vocals and mid-range instruments
• decrease long bass overtones in cymbals
• decrease guitar thinness
• lessen overly bright guitar overtones
• enhance background vocal breathiness
• disguise vocals and guitars that are slightly out of tune
• vocals: reduce "s" sound
CUT

## 5.5 EQ Setting Suggestions

Studio One Artist includes several EQ plugins. For an idea of where to start, check out the following generic EQ settings for several different instruments. As with the compression settings in ***Section 5.2.3***, the right EQ setting for any given instrument will depend upon the room and the tonality of the instrument.

### Vocals

#### Pop Female Vocals

| LOW ON/OFF | LOW SHELF | LOW FREQ (Hz) | LOW Q | LOW GAIN | LOW MID ON/OFF | LOW MID FREQ (Hz) | LOW MID Q | LOW MID GAIN |
|---|---|---|---|---|---|---|---|---|
| ON | OFF | 130 | 0.6 | -2 | ON | 465 | 0.6 | -2 |
| **HIGH MID ON/OFF** | **HI MID FREQ (kHz)** | **HIGH MID Q** | **HIGH MID GAIN** | **HIGH ON/OFF** | **HIGH SHELF** | **HIGH FREQ (kHz)** | **HIGH Q** | **HIGH GAIN** |
| ON | 2.4 | 0.4 | +2 | ON | OFF | 6.0 | 0.3 | +8 |

#### Rock Female Vocals

| LOW ON/OFF | LOW SHELF | LOW FREQ (Hz) | LOW Q | LOW GAIN | LOW MID ON/OFF | LOW MID FREQ (Hz) | LOW MID Q | LOW MID GAIN |
|---|---|---|---|---|---|---|---|---|
| ON | ON | 155 | N/A | +4 | ON | 465 | 0.4 | +6 |
| **HIGH MID ON/OFF** | **HI MID FREQ (kHz)** | **HIGH MID Q** | **HIGH MID GAIN** | **HIGH ON/OFF** | **HIGH SHELF** | **HIGH FREQ (kHz)** | **HIGH Q** | **HIGH GAIN** |
| ON | 1.4 | 0.6 | +6 | ON | OFF | 4.2 | 0.5 | +2 |

### Pop Male Vocals

| LOW ON/OFF | LOW SHELF | LOW FREQ (Hz) | LOW Q | LOW GAIN | LOW MID ON/OFF | LOW MID FREQ (Hz) | LOW MID Q | LOW MID GAIN |
|---|---|---|---|---|---|---|---|---|
| ON | OFF | 225 | 0.3 | -2 | ON | 960 | 0.3 | 0 |
| HIGH MID ON/ OFF | HI MID FREQ (kHz) | HIGH MID Q | HIGH MID GAIN | HIGH ON/OFF | HIGH SHELF | HIGH FREQ (kHz) | HIGH Q | HIGH GAIN |
| ON | 2.0 | 0.6 | +2 | ON | OFF | 7.2 | 0.5 | +4 |

### Rock Male Vocals

| LOW ON/OFF | LOW SHELF | LOW FREQ (Hz) | LOW Q | LOW GAIN | LOW MID ON/OFF | LOW MID FREQ (Hz) | LOW MID Q | LOW MID GAIN |
|---|---|---|---|---|---|---|---|---|
| ON | OFF | 155 | 0.5 | +2 | ON | 265 | 0.3 | -6 |
| HIGH MID ON/ OFF | HI MID FREQ (kHz) | HIGH MID Q | HIGH MID GAIN | HIGH ON/OFF | HIGH SHELF | HIGH FREQ (kHz) | HIGH Q | HIGH GAIN |
| ON | 2.4 | 0.6 | -2 | ON | ON | 7.2 | 0.6 | +4 |

## Percussion

### Snare

| LOW ON/OFF | LOW SHELF | LOW FREQ (Hz) | LOW Q | LOW GAIN | LOW MID ON/OFF | LOW MID FREQ (Hz) | LOW MID Q | LOW MID GAIN |
|---|---|---|---|---|---|---|---|---|
| ON | OFF | 130 | 0.6 | -4 | ON | 665 | 0.5 | +4 |
| HIGH MID ON/ OFF | HI MID FREQ (kHz) | HIGH MID Q | HIGH MID GAIN | HIGH ON/OFF | HIGH SHELF | HIGH FREQ (kHz) | HIGH Q | HIGH GAIN |
| ON | 1.6 | 0.3 | +4 | ON | ON | 4.2 | N/A | +4 |

### Left/Right (Stereo) Overheads

| LOW ON/OFF | LOW SHELF | LOW FREQ (Hz) | LOW Q | LOW GAIN | LOW MID ON/OFF | LOW MID FREQ (Hz) | LOW MID Q | LOW MID GAIN |
|---|---|---|---|---|---|---|---|---|
| ON | OFF | 108 | 0.6 | -2 | ON | 385 | 0.6 | -2 |
| HIGH MID ON/ OFF | HI MID FREQ (kHz) | HIGH MID Q | HIGH MID GAIN | HIGH ON/OFF | HIGH SHELF | HIGH FREQ (kHz) | HIGH Q | HIGH GAIN |
| ON | 2.9 | 0.3 | 0 | ON | ON | 8.0 | N/A | +4 |

### Kick Drum

| LOW ON/OFF | LOW SHELF | LOW FREQ (Hz) | LOW Q | LOW GAIN | LOW MID ON/OFF | LOW MID FREQ (Hz) | LOW MID Q | LOW MID GAIN |
|---|---|---|---|---|---|---|---|---|
| ON | OFF | 108 | 0.4 | +4 | ON | 265 | 2.0 | -4 |
| HIGH MID ON/ OFF | HI MID FREQ (kHz) | HIGH MID Q | HIGH MID GAIN | HIGH ON/OFF | HIGH SHELF | HIGH FREQ (kHz) | HIGH Q | HIGH GAIN |
| ON | 1.6 | 0.6 | 0 | ON | OFF | 6.0 | 2.0 | +4 |

## Fretted Instruments

### Electric Bass

| LOW ON/OFF | LOW SHELF | LOW FREQ (Hz) | LOW Q | LOW GAIN | LOW MID ON/OFF | LOW MID FREQ (Hz) | LOW MID Q | LOW MID GAIN |
|---|---|---|---|---|---|---|---|---|
| ON | ON | 36 | N/A | -8 | ON | 130 | 0.4 | +4 |
| HIGH MID ON/ OFF | HI MID FREQ (kHz) | HIGH MID Q | HIGH MID GAIN | HIGH ON/OFF | HIGH SHELF | HIGH FREQ (kHz) | HIGH Q | HIGH GAIN |
| ON | 2.0 | 0.6 | +4 | ON | ON | 4.2 | N/A | +1 |

### Acoustic Guitar

| LOW ON/OFF | LOW SHELF | LOW FREQ (Hz) | LOW Q | LOW GAIN | LOW MID ON/OFF | LOW MID FREQ (Hz) | LOW MID Q | LOW MID GAIN |
|---|---|---|---|---|---|---|---|---|
| ON | OFF | 155 | 0.4 | +4 | ON | 665 | 2.0 | +2 |
| HIGH MID ON/ OFF | HI MID FREQ (kHz) | HIGH MID Q | HIGH MID GAIN | HIGH ON/OFF | HIGH SHELF | HIGH FREQ (kHz) | HIGH Q | HIGH GAIN |
| ON | 2.0 | 0.3 | 0 | ON | ON | 6.0 | N/A | +4 |

### Distorted Electric Guitar

| LOW ON/OFF | LOW SHELF | LOW FREQ (Hz) | LOW Q | LOW GAIN | LOW MID ON/OFF | LOW MID FREQ (Hz) | LOW MID Q | LOW MID GAIN |
|---|---|---|---|---|---|---|---|---|
| ON | OFF | 320 | 0.5 | +6 | ON | 960 | 0.4 | 0 |
| HIGH MID ON/ OFF | HI MID FREQ (kHz) | HIGH MID Q | HIGH MID GAIN | HIGH ON/OFF | HIGH SHELF | HIGH FREQ (kHz) | HIGH Q | HIGH GAIN |
| ON | 3.5 | 1.0 | +4 | ON | ON | 12 | N/A | 0 |

## Keyboards

### Piano

| LOW ON/OFF | LOW SHELF | LOW FREQ (Hz) | LOW Q | LOW GAIN | LOW MID ON/OFF | LOW MID FREQ (Hz) | LOW MID Q | LOW MID GAIN |
|---|---|---|---|---|---|---|---|---|
| ON | ON | 108 | N/A | -2 | ON | 665 | 0.2 | +2 |
| HIGH MID ON/ OFF | HI MID FREQ (kHz) | HIGH MID Q | HIGH MID GAIN | HIGH ON/OFF | HIGH SHELF | HIGH FREQ (kHz) | HIGH Q | HIGH GAIN |
| ON | 2.9 | 0.4 | +2 | ON | OFF | 7.2 | 0.6 | +4 |

# 6.0 Technical Information

## 6.1 iOne and iTwo Technical Specifications

| | AudioBox iOne | AudioBox iTwo |
|---|---|---|
| **Microphone Preamp** | | |
| Type | XLR female, balanced | XLR female, balanced |
| Frequency Response (±3.0 dB) | 10 Hz to 40 kHz | 10 Hz to 40 kHz |
| Input Impedance (Balanced) | 1,200Ω | 1,200Ω |
| THD+N (unwtd, 1 kHz @ +4 dBu Output, Unity Gain) | < 0.008% | < 0.008% |
| S/N Ratio (Unity Gain, Ref. = +4 dBu, 20 Hz to 22 kHz) | > 95 dB | > 95 dB |
| Common Mode Rejection Ratio (1 kHz, 52 dB Gain) | > 65 dB | |
| Gain Control Range (± 1 dB) | 0 dB to 52 dB | 0 dB to 52 dB |
| Max Input Level (Unity Gain, 1 kHz @ 0.5% THD+N) | +10 dBu | +10 dBu |
| Phantom Power (±2 VDC) | +48 VDC | +48 VDC |
| **Line Input** | | |
| Type | ¼" TRS female, balanced | ¼" TRS female, balanced |
| Input Impedance | 10 kΩ | 10 kΩ |
| **Instrument Input** | | |
| Type | ¼" TS female, unbalanced | ¼" TS female, unbalanced |
| Input Impedance | 1 MΩ | 1 MΩ |
| **Line Outputs** | | |
| Type | ¼" TRS female, balanced | ¼" TRS female, balanced |
| Output Impedance | 51Ω | 51Ω |
| **Headphone Output** | | |
| Type | ¼" TRS active stereo | ¼" TRS active stereo |
| Maximum Output | 60 mW/ch. @ 60Ω Load | 60 mW/ch. @ 60Ω Load |
| Frequency Response (±1.0 dB) | 20 Hz - 30 kHz | 20 Hz - 30 kHz |
| **MIDI I/O** | | |
| Type | Dual 5-pin female DIN | Dual 5-pin female DIN |
| **Digital Audio** | | |
| ADC Dynamic Range (A-wtd, 48 kHz Sample Rate) | 105 dB | 105 dB |
| DAC Dynamic Range (A-wtd, 48 kHz Sample Rate) | 105 dB | 105 dB |
| Bit Depth | 24 | 24 |
| Reference Level for 0 dBFS | +10 dBu | +10 dBu |
| Internal Sample Frequency Selections (kHz) | 44.1, 48, 88.2, 96 | 44.1, 48, 88.2, 96 |
| Power | USB bus-powered | USB bus-powered |
| **Physical** | | |
| Height | 1.71" (43.5 mm) | 1.71" (43.5 mm) |
| Depth | 5.31" (135 mm) | 5.31" (135 mm) |
| Width | 5.91" (150 mm) | 7.56" (192 mm) |
| Weight | 1.35 lbs (0.62 kg) | 1.35 lbs (0.62 kg) |

# 7.0 Troubleshooting and Warranty

## 7.1 Troubleshooting

Many technical issues can arise when using a standard computer as a digital audio workstation (DAW). PreSonus can only provide support for issues that directly relate to the AudioBox iOne and iTwo interface and Studio One digital audio workstation software. PreSonus does not provide support for computer hardware, operating systems, and non-PreSonus hardware and software, and it may be necessary to contact the manufacturer of these products for technical support.

Please check the PreSonus Web site (www.presonus.com) regularly for software information and updates, firmware updates, and support documentation, including frequently asked questions.

Online technical support is available at www.presonus.com/support/Contact-Technical-Support.

Technical support is available via email at techsupport@presonus.com.

PreSonus telephone technical support is available to customers in the USA on Monday through Friday from 9 a.m. to 5 p.m. Central Time by calling 1-225-216-7887. Customers outside of the USA should contact their national or regional distributor for telephone technical support. A list of international distributors is provided at www.presonus.com/buy/international_distributors.

**AudioBox Will Not Connect to Computer**

Verify that the USB cable is properly connected both to the AudioBox and to your computer. Disconnect unnecessary peripheral USB devices.

Verify that your AudioBox is connected to a USB connection.

**AudioBox Will Not Connect to iPad**

Verify that the USB cable is properly connected both to the AudioBox and to your a compatible 5V, 2.1A (2,100 mA) USB power supply, such as the one that came with the iPad.

Verify that your AudioBox is connected to the iPad using an Apple 30-pin or Lightning cable.

**Input Phasing While Monitoring Using Studio One or Another DAW**

Verify that the mix knob on the front of the AudioBox is turned fully clockwise to the Playback position. If you prefer to use the analog zero-latency monitoring on your AudioBox, make sure that the input channel in your DAW has software monitoring disabled. You will be monitoring through the analog bus on your AudioBox, not your DAW, using the Mix knob on the front of your AudioBox to blend the playback from your computer with your analog input signal.

## 7.2 Warranty

PreSonus's warranty obligations for this hardware product are limited to the terms set forth below:

**How Consumer Law Relates To This Warranty:**

THIS WARRANTY GIVES YOU SPECIFIC LEGAL RIGHTS, AND YOU MAY HAVE OTHER RIGHTS THAT VARY FROM STATE TO STATE (OR BY COUNTRY OR PROVINCE). OTHER THAN AS PERMITTED BY LAW, PRESONUS DOES NOT EXCLUDE, LIMIT OR SUSPEND OTHER RIGHTS YOU MAY HAVE, INCLUDING THOSE THAT MAY ARISE FROM THE NONCONFORMITY OF A SALES CONTRACT. FOR A FULL UNDERSTANDING OF YOUR RIGHTS YOU SHOULD CONSULT THE LAWS OF YOUR COUNTRY PROVINCE OR STATE.

**PreSonus Products And EU Statutory Warranty:**

When you purchase PreSonus products, European Union consumer law provides statutory warranty rights in addition to the coverage you receive from the PreSonus limited warranty. A summary of the EU Statutory Warranty and the PreSonus Limited Warranty is below:

| | EU Consumer Law | PreSonus Limited Warranty |
|---|---|---|
| Repair or Replacement Coverage For | Defects present when customer takes delivery | Defects arising after customer takes delivery |
| Warranty Period | 2 years (minimum) from original date of purchase (unless superseded by PreSonus) | 1 year from original date of purchase (unless superseded by PreSonus) |
| Cost of Coverage | Provided at no additional cost | Included at no additional cost |
| Who to contact to make a claim | The seller | PreSonus technical support for your region |

**What This Warranty Covers:**

PreSonus Audio Electronics, Inc., ("PreSonus") warrants defects in material and workmanship in PreSonus-branded products under normal use. This Limited Warranty applies only to hardware products manufactured by or for PreSonus that can be identified by the PreSonus trademark, trade name, or logo affixed to them.

**Exclusions and Limitations:**

This warranty does not cover the following:

1. Damage caused by accident, abuse, improper installation, failure to follow instructions in the applicable owner's manual or improper operation, rental, product modification, alteration, or neglect.
2. Damage from improper grounding, faulty wiring (AC and signal), faulty equipment, or connection to a voltage range outside published specifications (see applicable owner's manual).
3. Damage to drivers or diaphragm assemblies found to have burnt voice coils from over/under driving or signal surge from another device.
4. Damage occurring during shipment or improper handling.
5. Damage caused by repair or service performed by persons not authorized by PreSonus.
6. Products on which the serial number has been altered, defaced, or removed.
7. Products purchased from an un-authorized PreSonus dealer (products that have transferable warranties are excluded from this provision provided the customer and the product are registered with PreSonus).

**Who This Warranty Protects:**

This Warranty protects only the original retail purchaser of the product (products that have transferable warranties are excluded from this provision provided the customer and the product are registered with PreSonus)

**How Long This Warranty Lasts:**

The Warranty begins on the original date of purchase from the retail purchaser and the duration is as follows:

| 1-Year Limited Warranty | | |
|---|---|---|
| **Product Category** | **Model** | **Transferable** |
| Recording Interfaces | AudioBox iOne, AudioBox iTwo, AudioBox Stereo, AudioBox Studio, AudioBox USB, AudioBox VSL (1818, 44, 22), FireStudio Project, FireStudio Mobile, FireStudio Mobile Studio, Music Creation Suite | No |
| Pre-amplifiers | ADL600, ADL700, BlueTube DP V2, DigiMax D8, Eureka, RC500, StudioChannel, TubePre V2 | No |
| StudioLive Mixers | SL-1642, SL-2442, SL-1602, SLM16.4.2 AI, SLM24.4.2 AI, SLM32.4.2AI | No |
| Monitoring & Controlling | Eris, Central Station Plus, FaderPort, HP4, HP60, Monitor Station, Monitor Station V2, Sceptre, Temblor | No |
| Signal Processing | ACP88 | No |
| Accessories | Covers, Dolly, PRM1 mic, Sub Pole, Breakout Cables, Power Supplies, M10 Kit | No |
| **3-Years Limited Warranty** | | |
| **Product Category** | **Model** | **Transferable** |
| Live Sound | StudioLive AI 328, 312, 315, S18 | Yes |

**What PreSonus Will Do:**

PreSonus will repair or replace, at our sole and absolute option, products covered by this warranty at no charge for labor or materials. If the product must be shipped to PreSonus for warranty service, the customer must pay the initial shipping charges. PreSonus will pay the return shipping charges.

**How to Get Warranty Service (USA):**

1. You must have an active user account with PreSonus and your hardware must be on file with your account. If you do not have an account, please go to: http://www.presonus.com/registration and complete the registration process.

2. Contact our Technical Support Department at (225) 216-7887 or log a support ticket at: http://support.presonus.com. TO AVOID THE POSSIBILITY OF SENDING IN A PRODUCT THAT DOES NOT HAVE A PROBLEM, ALL SERVICE REQUESTS SHALL BE CONFIRMED BY OUR TECH SUPPORT DEPARTMENT.

3. The return authorization number as well as shipping instructions shall be provided after your service request is reviewed and confirmed.

4. The product should be returned for service in the original product packaging. Products may be shipped in a manufactured "flight" or "road" style cases but PreSonus will NOT cover any shipping damage to these cases. Products that are not shipped in the original product package or a manufactured case may not receive a warranty repair, at PreSonus's sole discretion. Depending on the product model and the condition of your original packaging, your product may not be returned to you in the original packaging. The return shipping box may be a generic box that has been fitted for that model tested if the original gift box is not available.

**How to Get Warranty Service (outside of USA):**

1. You must have an active user account with PreSonus and your hardware must be on file with your account. If you do not have an account, please go to: http://www.presonus.com/registration and complete the registration process.

2. Contact the Technical Support/Service Department for your region at http://www.presonus.com/buy/international_distributors and follow procedures provided by your PreSonus contact.

**Limitation of Implied Warranties:**

ANY IMPLIED WARRANTIES, INCLUDING WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE, ARE LIMITED IN DURATION TO THE LENGTH OF THIS WARRANTY.

Some states, countries, or provinces do not allow limitations on how long an implied warranty lasts, so the above limitation may not apply to you.

**Exclusion of Damages:**

PRESONUS'S LIABILITY FOR ANY DEFECTIVE PRODUCT IS LIMITED TO THE REPAIR OR REPLACEMENT OF THE PRODUCT, AT PRESONUS'S SOLE OPTION. IF PRESONUS ELECTS TO REPLACE THE PRODUCT, THE REPLACEMENT MAY BE A RECONDITIONED UNIT. IN NO EVENT WILL PRESONUS BE LIABLE FOR DAMAGES BASED ON INCONVENIENCE, LOSS OF USE, LOST PROFITS, LOST SAVINGS, DAMAGE TO ANY OTHER EQUIPMENT OR OTHER ITEMS AT THE SITE OF USE, AND, TO THE EXTENT PERMITTED BY LAW, DAMAGES FOR PERSONAL INJURY, OR ANY OTHER DAMAGES WHETHER INCIDENTAL, CONSEQUENTIAL OR OTHERWISE, EVEN IF PRESONUS HAS BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES.

Some states, countries, or provinces do not allow limitations on how long an implied warranty lasts, so the above limitation may not apply to you.

If you have any questions about this warranty or service received, please contact PreSonus (USA) at (225) 216-7887 or one of our authorized international distributors at: http://www.presonus.com/buy/international_distributors.

**Product features, design, and specifications are subject to change without notice.**

# Declaration of Conformity

Responsible Party: **PreSonus Audio Electronics**

Address: **18111 Grand Bay Ct., Baton Rouge, LA 70806 USA**

Phone: **225-216-7887**

declares that **AudioBox™ iOne and iTwo** comply with Part 15 of the FCC rules.

Operation is subject to the following two conditions:

1. This device may not cause harmful interference, and;
2. This device must accept any interference received, including interference that may cause undesired operation

**Note:** No product support is available when you call the number above. Refer to your Certificate of Warranty in your Owner's Manual for PreSonus' Product Support telephone number.

PreSonus

**Baton Rouge • USA • www.presonus.com**

# Added bonus: PreSonus' previously Top Secret recipe for. . .

# Chicken and Andouille Gumbo

**Ingredients:**

- 1 C All-Purpose flour
- ¾ C Vegetable Oil
- 1 large onion (diced)
- 1 small onion (quartered)
- 6 celery stalks (diced)
- 1 large green bell pepper (diced)
- 3 cloves garlic (2 minced, 1 whole)
- 1 lb link Andouille sausage
- 4 Chicken leg quarters
- 4 qt water
- 4 bay leaves
- 1 tsp thyme
- 1 tsp Old Bay seasoning
- 1-2 C frozen okra, sliced
- ¼ C fresh parsley, minced
- 6-8 eggs (optional)

**Cooking Instructions:**

1. In a large pot, combine whole chicken leg quarters, water, quartered onion, Old Bay, 2 bay leaves and 1 whole clove garlic. Cover and bring to a low boil. Simmer stock until chicken is falling off the bone. Remove the chicken and set aside. Discard the onion, bay leaves, and garlic, reserving the liquid.
2. In a heavy saucepan, heat 1 Tbsp of the oil on medium high heat and brown the andouille until it is cooked through. Set aside sausage for later.
3. In the same saucepan, add and heat remaining oil. Slowly add flour 1-2 Tbsp at a time, stirring continuously. Continue cooking and stirring the roux until it is a dark brown (it should look like melted dark chocolate). Be careful to not to get the oil too hot or the flour will burn and you'll have to start over.
4. Once roux has reached the correct color, add diced onion, celery, green pepper, and minced garlic. Cook until vegetables are very tender. Do not cover.
5. Slowly add 1 quart of chicken broth and bring to a low boil, stirring constantly.
6. Transfer roux mixture to a soup pot and bring to low boil. Do not cover, the roux will settle on the bottom of the pot and burn.
7. Add remaining chicken broth, bay leaves, and thyme. Simmer for 30 minutes.
8. While gumbo is simmering, debone and shred chicken and slice the andouille.
9. Add chicken and andouille to gumbo and return to a simmer. Simmer for 30-45 minutes.
10. Stir in frozen okra and parsley and bring to a rolling boil.
11. **Optional:** Crack one egg into a teacup and quickly pour into the boiling gumbo. Repeat with the other eggs being careful not to cluster them too closely. After all the eggs have risen back to the surface, reduce heat and simmer.
12. Correct seasoning with salt and pepper (red, white and/or black) if necessary.
13. Serve over rice with potato salad.

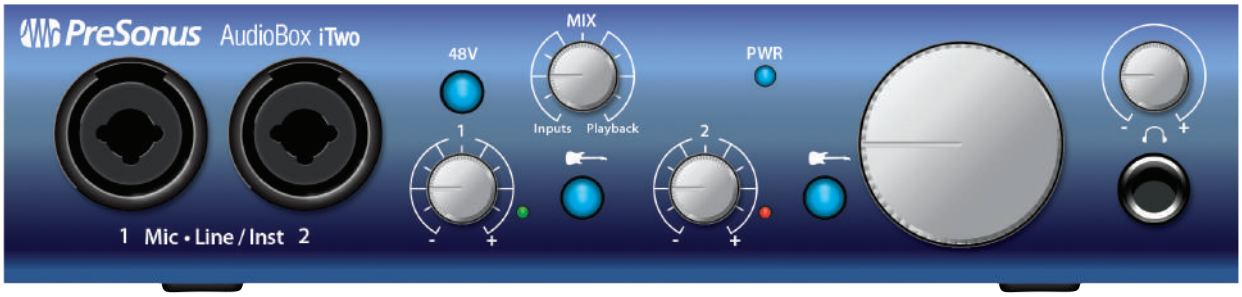
**Serves 12**

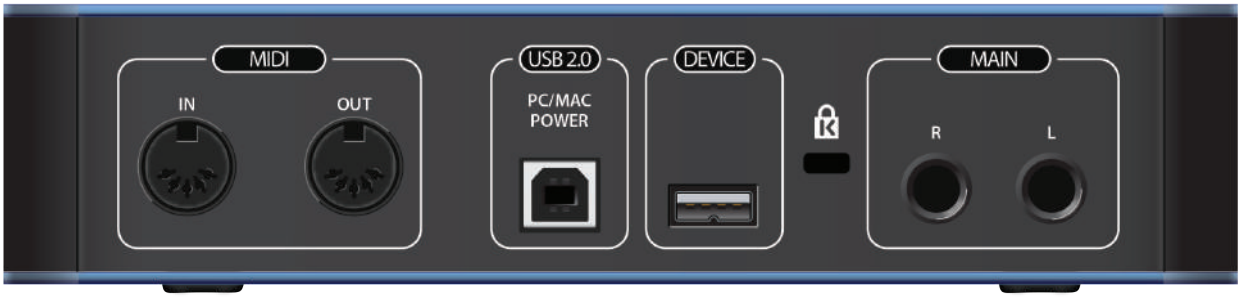
© 2014 PreSonus Audio Electronics, Inc. All Rights Reserved. AudioBox, Ceres, CoActual, DigiMax, Eris, FireStudio, Nimbit, PreSonus, QMix, Riff to Release, Sceptre, Temblor, StudioLive, and XMAX are trademarks or registered trademarks of PreSonus Audio Electronics, Inc. Capture, Impact, Mixverb Presence, RedLightDist, SampleOne, Studio One, and Tricomp are trademarks or registered trademarks of PreSonus Software Ltd. Mac and Mac OS are registered trademarks of Apple, Inc., in the U.S. and other countries. Windows is a registered trademark of Microsoft, Inc., in the U.S. and other countries. Temporal EQ and TQ are trademarks of Fulcrum Acoustic. Other product names mentioned herein may be trademarks of their respective companies. All specifications subject to change without notice...except the recipe, which is a classic.

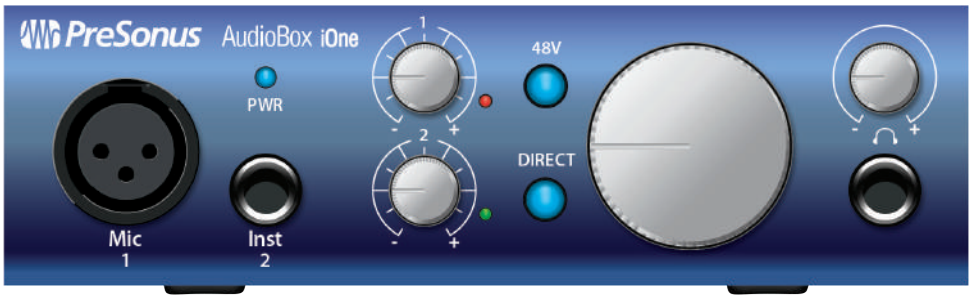
# AudioBox™ iOne and iTwo

## USB/iPad Recording Systems

Owner's Manual



PreSonus®

18011 Grand Bay Ct. • Baton Rouge,
Louisiana 70809 USA• 1-225-216-7887
**www.presonus.com**

Part# 820-AB0043-B